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October 18, 2016

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The Honorable Jocelyn D. Boyd
Chief Clerk/Administrator
Public Service Commission of South Carolina
101 Executive Center Drive
Columbia, SC 29210

Re: Petition of the South Carolina Telephone Coalition to Require
Interconnected Voice Over Internet Protocol ("Interconnected VoIP")
Service Providers to Contribute to the South Carolina Universal Service
Fund
Docket No. 2016-267-C

Dear Ms. Boyd:

Attached for filing on behalf of the South Carolina Telephone Coalition
("SCTC") please find the Direct Testimony of Douglas Meredith in the above
referenced docket.

Thank you for your assistance in this matter.

Very truly yours,

McNAIR LAW FIRM, P.A.



Margaret M. Fox

MMF:khh

Enclosures

cc: All counsel of record (w/Encls.)

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BEFORE
THE PUBLIC SERVICE COMMISSION OF SOUTH CAROLINA
DOCKET NO. 2016-267-C

IN RE:)	
Petition of the South Carolina Telephone)	CERTIFICATE OF SERVICE
Coalition to Require Interconnected Voice)	
Over Internet Protocol ("Interconnected)	
VoIP") Service Providers to Contribute to)	
the South Carolina Universal Service Fund)	
<hr/>		

This is to certify that I, Kathy H. Handrock, a Paralegal with the McNair Law Firm, P.A., have this date served one (1) copy of the Direct Testimony of Douglas Meredith on Behalf of the South Carolina Telephone Coalition, Docket No. 2016-267-C in the above-referenced matter to the person(s) named below by causing said copy to be deposited in the United States Postal Service, first class postage prepaid and affixed thereto, and addressed as shown below:

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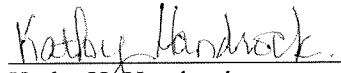
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October 18, 2016

Columbia SC

BEFORE
THE PUBLIC SERVICE COMMISSION OF
SOUTH CAROLINA

Docket No. 2016-267-C

In Re: Petition of the South Carolina Telephone Coalition)
To Require Interconnected Voice over Internet Protocol)
("Interconnected VoIP") Service Providers to Contribute)
To the South Carolina Universal Service Fund)
_____)

DIRECT TESTIMONY OF
DOUGLAS MEREDITH
ON BEHALF OF
THE SOUTH CAROLINA
TELEPHONE COALITION

October 18, 2016

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List of Exhibits

- Exhibit DDM-1 — Douglas Meredith Curriculum Vitae
- Exhibit DDM-2 — List of SCTC Members
- Exhibit DDM-3 — VoIP and Unified Communications Define the Future
- Exhibit DDM-4 — JSI Interconnected VoIP Regulatory Obligations
- Exhibit DDM-5 — Excerpt from NRRI Report No. 15-05: State Universal Service Funds
2014

1 **I. Introduction**

2 **Q: Please state your full name, address, place of employment and position with your**
3 **employer.**

4 A: My full name is Douglas Duncan Meredith. I am employed by John Staurulakis, Inc.
5 ("JSI") as Director – Economics and Policy. My office is located at 547 Oakview Lane,
6 Bountiful, Utah 84010.

7
8 **Q: What is JSI and what does it do?**

9 A: JSI is a telecommunications consulting firm headquartered in Greenbelt, Maryland. JSI
10 has provided telecommunications consulting services to local exchange carriers since
11 1963.

12
13 **Q: Please describe your professional experience and educational background prior to**
14 **coming to work for JSI.**

15 A: Prior to my work at JSI, I was an independent research economist in the District of
16 Columbia and a graduate student at the University of Maryland – College Park. I have a
17 Bachelor of Arts degree in economics from the University of Utah, and a Masters degree
18 in Economics from the University of Maryland – College Park. While attending the
19 University of Maryland – College Park, I was also a Ph.D. candidate in Economics, having
20 completed all coursework, comprehensive and field examinations for a Doctorate of
21 Economics.

23 **Q: Please describe your duties and responsibilities with JSI.**

24 A: As the Director of Economics and Policy at JSI, I assist clients with the development of
25 policy pertaining to economics, pricing and regulatory affairs. I have been employed by
26 JSI since 1995.

27
28 In my employment at JSI, I have participated in numerous proceedings for rural and non-
29 rural telephone companies. These activities include, but are not limited to, the creation of
30 forward-looking economic cost studies, the development of policy related to the
31 application of the rural safeguards for qualified local exchange carriers, the determination
32 of Eligible Telecommunications Carriers, the sustainability and application of universal
33 service policy for telecommunication carriers, as well as supporting incumbent local
34 exchange carriers in arbitration proceedings and rural exemption and suspension and/or
35 modification proceedings.

36
37 In addition to assisting telecommunications carrier clients, I have served as the economic
38 advisor for the Telecommunications Regulatory Board of Puerto Rico (“Puerto Rico Board
39 of Commissioners”) since 1997. In this capacity, I provide economic and policy advice to
40 the Puerto Rico Board of Commissioners on telecommunications issues that have either a
41 financial or economic impact on carriers or end-users. I have participated in a number of
42 arbitration panels established by the Puerto Rico Board of Commissioners to arbitrate
43 interconnection issues under Section 252 of the Telecommunications Act of 1996.

I have participated in numerous national incumbent local exchange carrier and telecommunications groups, including those headed by NTCA, OPASTCO, USTelecom, and the Rural Policy Research Institute. My participation in these groups focuses on the development of policy recommendations for advancing universal service and telecommunications capabilities in rural communities and other policy matters.

Q: Have you testified previously in federal and state regulatory proceedings on telecommunications issues?

A: Yes. In addition to testifying before the Public Service Commission of South Carolina (“PSC” or “Commission”) on multiple occasions, I have testified live or in pre-filed regulatory testimony in various states including Maine, Vermont, New Hampshire, New York, Michigan, Wisconsin, North Dakota, South Dakota, Texas, Tennessee, Colorado, Kentucky, and Utah. I have also assisted clients in regulatory proceedings in many other states that did not require formal testimony, including Florida, Louisiana, Mississippi, Puerto Rico and Virginia. In addition to participation in state regulatory proceedings, I have assisted clients in federal regulatory proceedings in the filing of formal comments in various proceedings and submission of economic reports in an enforcement proceeding.

Q: Do you have a *curriculum vitae* that describes your education and work experience, and identifies the proceedings in which you have provided testimony or otherwise assisted?

A: Yes. It is identified as Exhibit DDM-1.

Q: On whose behalf are you testifying in this proceeding?

A: Today I am submitting testimony for the South Carolina Telephone Coalition (“SCTC”) and its member companies which are listed on Exhibit DDM-2.

Q: What is the purpose of your testimony?

A: My testimony is in support of the SCTC Petition to require interconnected voice over Internet Protocol (“interconnected VoIP”) service providers to contribute to the South Carolina Universal Service Fund (“SCUSF”) in a manner consistent with other providers of voice telecommunications service (“voice telephony”). My testimony includes the following sections:

(1) a brief historical background of interconnected VoIP;

(2) a review of the SCTC Petition; and,

(3) a discussion of my analysis in support of the Petition.

I have numerous exhibits that support and provide additional information related to the SCTC proposal that are identified in the List of Exhibits.

II. Background of Interconnected VoIP Service

Q: What is interconnected VoIP service?

A: Interconnected VoIP is a voice service offered by a provider to end-user customers. Perhaps the best way to explain interconnected VoIP is to compare it to traditional circuit switched voice service offered by incumbent local exchange carriers. Traditional circuit switched voice service allows an end-user (calling party) to initiate a two-way path to a called party when both parties—the calling and called parties—are connected to the public

switched network. Both the calling party and called party are end-users of a telecommunications service offered by either an incumbent local exchange carrier or a competitive local exchange carrier.

Interconnected VoIP service is similar to the traditional circuit switched voice service I just described. An interconnected VoIP provider offers a service to end-user customers that enables real-time, two-way voice communications. To use an interconnected VoIP service there is a requirement that the subscriber has a broadband connection capable of running an interconnected VoIP session. Furthermore, the subscriber needs to have customer premises equipment (“CPE”) capable of running an Internet Protocol session.

Q: Help me unpack what you just described. Is the traditional circuit switched voice service you described the service traditionally understood as local exchange service offered by SCTC members?

A: Yes. Before interconnected VoIP service became commercially available, SCTC members as well as all other incumbent and competitive local exchange carriers offered some form of traditional circuit switched voice service.

Q: Does a traditional circuit switched voice technology require the dedicated use of a local loop connected to an end-office switch?

A: Yes. The local loop provided to the subscriber is a dedicated facility upon which a local switch announces a dial-tone when the end-user subscriber initiates a call—*i.e.*, picks up the handset or activates a speaker/microphone feature.

113 **Q: Do SCTC members offer traditional circuit switched voice services?**

114 A: Yes. In addition, SCTC members have begun to offer an interconnected VoIP service to
115 end-user customers. Many in the industry have observed that interconnected VoIP service
116 is a replacement for traditional circuit switched voice service. (*See Exhibit DDM-3 —*
117 *VoIP is the Future*) Often when the future of communications is described, VoIP plays a
118 vital role for SCTC members and other providers in the evolution of the traditional circuit
119 switched voice service as a communications solution for end-user subscribers.

120

121 **Q: Are you suggesting that VoIP service will be used increasingly to offer voice service**
122 **in the future?**

123 A: Yes. Due to a number of technical and economic considerations, interconnected VoIP
124 service will be used to offer voice service by SCTC members and other carriers.

125

126 **Q: Does interconnected VoIP service require a dedicated loop to the end-user subscriber**
127 **to be used only for the voice service?**

128 A: No. VoIP service does not require a dedicated circuit switched loop. Instead
129 interconnected VoIP requires the user to have a robust broadband connection to initiate a
130 voice service session.

131

132 **Q: Please describe what type of broadband is required for interconnected VoIP.**

133 A: The Federal Communications Commission (“FCC”) discusses two minimum
134 requirements: speed and latency. In today’s marketplace, speed is generally not a factor
135 in determining whether an end-user subscriber can initiate a VoIP session (*i.e.*, send or

136 receive a voice call using IP). Latency, however, is an important consideration to enable
137 two-way simultaneous voice service. For a VoIP session, the FCC suggests that latency
138 generally needs to be less than 100 milliseconds to enable acceptable two-way voice
139 communication. More often than not, a basic broadband service offered by providers in
140 today's marketplace is capable of enabling interconnected VoIP service.

141
142 **Q: In addition to a broadband connection, is IP-enabled CPE required for VoIP service?**

143 A: Yes. IP-enabled CPE is required to receive IP communications. This can be a specialized
144 handset, a router, or conversion device that enables the use of traditional handsets. These
145 are very common types of CPE available in the marketplace today.

146
147 **Q: Please describe what is meant by the adjective "interconnected" when it is used to**
148 **describe VoIP.**

149 A: Interconnected VoIP is a service whereby a subscriber may initiate or receive a call using
150 the public switched network. (The public switched network allows all providers and
151 carriers to interconnect, thereby enabling end-users to call VoIP subscribers as well as
152 traditional circuit switched voice service subscribers.) Describing a provider as
153 "interconnected" indicates that the provider's end-user customers are able to originate calls
154 or terminate calls using the public switched network and call to or receive calls from all
155 other parties similarly interconnected—which includes the universe of subscribers who
156 have a working telephone number.

158 **Q: Does an end-user subscribing to an interconnected VoIP service receive a telephone**
159 **number to send and receive telephone calls?**

160 A: Yes. I don't know of any instance where an end-user subscribing to an interconnected
161 VoIP service doesn't receive a new or use an existing telephone number consistent with
162 the North American Numbering Plan guidelines (NPA-NXX-XXXX) to send and receive
163 calls. The phone number is paired with an IP address that identifies the subscriber's
164 interconnection device. The ability to have a telephone number is the primary reason to
165 have an interconnected VoIP service. This feature contrasts sharply from non-
166 interconnected VoIP service where an IP address or name is used to identify the called and
167 calling parties' devices.

168
169 **Q: Do the terms "public switched network" and "public switched telephone network"**
170 **refer to the same network?**

171 A: The traditional circuit switched voice service referenced to the public switched telephone
172 network. However, recently the FCC has changed the definition of the public switched
173 network to include IP addresses as well as telephone numbers. (*See* Open Internet Order,
174 30 FCC Rcd. at 5610, *aff'd*, U.S. Telecom Ass'n v. FCC, 64 CR 1663 (D.C. Cir. 2016))
175 Thus, the public switched network is more expansive than the traditional public switched
176 telephone network. This change shows how the FCC's policies are evolving to include IP
177 services.

178
179 **Q: Do the Federal Regulations define interconnected VoIP service?**

180 A: Yes. 47 C.F.R. § 9.3 states that interconnected VoIP:

- 181 (1) Enables real-time, two-way voice communications;
182 (2) Requires a broadband connection from the user's location;
183 (3) Requires Internet protocol-compatible customer premises equipment (CPE);
184 and
185 (4) Permits users generally to receive calls that originate on the public switched
186 telephone network and to terminate calls to the public switched telephone
187 network.
188

189 The SCTC Petition describes interconnected VoIP service as a service whereby an end-
190 user subscriber may initiate or receive two-way voice communications (calls or sessions)
191 from all points interconnected on the public switched telephone network.

192

193 **Q: Are providers of interconnected VoIP service in South Carolina certificated by the**
194 **Commission?**

195 A: I understand that there are two types of interconnected VoIP service providers in South
196 Carolina. Some providers have sought a certificate from the Commission to provide
197 interconnected VoIP service, and others have not.

198

199 **Q: Is the result of the SCTC Petition that all "interconnected VoIP" service providers**
200 **and traditional local exchange carriers would contribute to the South Carolina USF**
201 **on the same basis?**

202 A: Yes. Currently some providers that offer interconnected VoIP services do not contribute
203 while others do contribute. The SCTC Petition would require all parties who provide
204 interconnected VoIP services using the public switched telephone network to contribute to
205 the SCUSF.

206

FCC Treatment of Interconnected VoIP Service

Q: How does the FCC regulate interconnected VoIP service providers?

A: The FCC has a history of regulating interconnected VoIP service providers. The FCC has declined to classify interconnected VoIP service as a telecommunications service, thereby permitting a light regulatory touch to be applied to this emerging service. (*See In Re IP-Enabled Services*, NPRM, 19 FCC Rcd 4863, 4876 (2004)) Over time, the FCC has increased regulations on interconnected VoIP service providers. Today, interconnected VoIP service providers have many regulations that apply to traditional local exchange carriers, including the requirement to contribute to the Federal Universal Service Fund. JSI provides regulatory compliance services for interconnected VoIP service providers and Exhibit DDM-4 lists the current FCC requirements for such providers. The requirements include 911 service, CALEA, CPNI, Form 499 submissions, Federal USF contributions, contributions to interstate TRS, LNP, disability access, numbering, Form 477 submissions, regulatory fees, discontinuance, outage reporting, call blocking prohibitions, access charges, international reporting, and power backup requirements. These requirements are substantial and mirror some of the requirements of local exchange carriers even though interconnected VoIP isn't classified as a telecommunications service for federal regulatory purposes.

III. Review of SCTC Petition

Q: Please describe the SCTC Petition.

A: The SCTC Petition consists of a preamble, 17 numbered paragraphs and two requests. Paragraphs 1 and 2 identify SCTC and its members as local exchange carriers that contribute to the SCUSF.

Q: Does paragraph 3 identify the state statute that informs your analysis in this proceeding?

A: Yes. I understand that the state statute established the SCUSF to be consistent with federal policies in the provision of what is described as basic local exchange telephone service. Federal policy requires interconnected VoIP service providers to contribute to the federal USF, suggesting that the FCC considers interconnected VoIP service to be similar enough to traditional voice service that both services should be treated in a similar manner in the context of Universal Service support. Since SCTC members are using both technologies—traditional circuit switched and interconnected VoIP—to offer basic local exchange service to end-user customers, this federal policy strongly supports treating the entities offering these services the same for purposes of SCUSF contributions.

Q: Paragraph 4 identifies another section of the statute, followed by paragraphs 6 through 8. How should these statutory and court references inform the Commission in this proceeding?

A: The Commission should require interconnected VoIP service providers to contribute to SCUSF consistent with how the FCC has required similar federal USF contributions without specifying the regulatory classification of such services.

Q: The SCTC Petition asks the Commission to find that all interconnected VoIP service providers, regardless of whether they hold a Certificate of Public Convenience and Necessity, must contribute to the SCUSF and those that aren't currently contributing are required to do so on a prospective basis. Is this recommendation consistent across all interconnected VoIP service providers?

A: Yes. It shouldn't matter whether an interconnected VoIP service provider has a certificate or not. The possession of a certificate does not alter the service these providers offer to the public. Each service enables end-user customers to benefit from a physical connection to the public switched telephone network built and maintained in part with the disbursement of SCUSF support. All benefit from being interconnected to the public switched telephone network and all should contribute.

IV. Analysis in Support of Petition

Q: What analysis have you performed to reach your recommendation in this proceeding?

A: I have examined three issues or policies that support the SCTC Petition. These are (1) the SCTC request is competitively neutral; (2) the SCTC request is tailored to promote the health and future stability of the SCUSF; and, (3) the benefits of a network are shared by all who interconnect to the public switched telephone network in South Carolina and should be supported by all who provide interconnected services.

Competitively Neutral

Q: Is the requirement that all interconnected VoIP service providers contribute to the SCUSF a competitively neutral policy?

A: Yes. The requirement that all providers of interconnected voice services offered to end-users in South Carolina contribute to the SCUSF is competitively neutral because it applies a uniform USF support policy to all voice service providers and carriers in the state.

Q: Why should the contribution requirement apply to interconnected VoIP service?

A: From a customer's viewpoint, voice service from an interconnected VoIP provider is similar to traditional circuit switched voice service. Comparing the CPE for traditional circuit switched service and interconnected VoIP service, the only difference is the use of a device that is internal or external to a handset that allows for voice calls to be sent or received over the end-user's broadband service. I have been an end-user customer of both services, and I have used the services in a similar manner to make and receive voice calls.

286 **Q: Do the two services compete against one another?**

287 A: Yes. In the SCTC markets traditional circuit switched voice service competes with
288 interconnected VoIP services. I also note that certificated interconnected VoIP service
289 providers also compete with non-certificated interconnected VoIP service providers in
290 South Carolina. In this latter instance, the services are perfect substitutes to each other—
291 they both use the same CPE and require the use of a broadband connection. In all of these
292 instances, it is in the public interest to have a competitively neutral contribution policy that
293 requires all interconnected voice service providers to contribute to the SCUSF.

294

295 **Health and Future Stability of the SCUSF**

296 **Q: How is the requirement that all interconnected VoIP service providers contribute to**
297 **the SCUSF ensure the health and future stability of SCUSF?**

298 A: SCTC members and other providers are migrating to a VoIP platform for voice service as
299 the technology evolves. The Commission has the opportunity to ensure the continued
300 availability of affordable voice service for all in South Carolina by making the contribution
301 base as broad as possible by requiring a contribution from all interconnected voice service
302 providers.

303

304 **Benefits of a Network**

305 **Q: You have stated that all interconnected voice services benefit from being**
306 **interconnected in the public switched telephone network. Is this a basis for the**
307 **Commission to establish a uniform contribution policy?**

308 A: Yes. All interconnected voice service providers benefit from a robust South Carolina local
309 exchange network, and all should equitably contribute into the SCUSF to maintain the
310 network enjoyed by all.

311
312 **Q: Do other states require interconnected VoIP service providers to contribute to their**
313 **state USF programs?**

314 A: Many states require contributions from VoIP service providers. Exhibit DDM-5 is an
315 excerpt of a NRRI report that describes the various state programs and what types of
316 entities contribute to each state fund. While many states require VoIP service providers to
317 contribute, other states have not changed their rules. So the best way to describe the activity
318 in other states is that it is in flux, with many states addressing the very issues raised in this
319 proceeding and concluding that interconnected VoIP service providers should be required
320 to contribute to their state USF programs.

321 **V. Conclusion**

322 **Q: Please review your recommendation to the Commission in this proceeding.**

323 A: For the reasons stated above, I recommend the Commission find that all interconnected
324 VoIP service providers, regardless of whether they hold a Certificate of Public
325 Convenience and Necessity, must contribute to the SCUSF and those that aren't currently

326 contributing are required to do so on a prospective basis. This finding is consistent with
327 the federal policy that requires interconnected VoIP service providers to contribute to the
328 federal universal service programs. By making this decision, the Commission will be
329 preserving and advancing the goals of the SCUSF in ensuring the future availability of
330 support for voice services offer throughout South Carolina.

331
332 **Q: Does this conclude you pre-filed direct testimony?**

333 **A:** Yes.

DDM-1

Douglas Duncan Meredith

547 South Oakview Lane
Bountiful, Utah 84010

EDUCATION:

Ph.D. Candidate: 1994, University of Maryland - College Park
Field areas: Advanced Macroeconomics and Monetary Economics.
(Ph.D. Candidacy requires successful completion of all Ph.D. coursework,
Comprehensive and Field examinations. Dissertation not completed.)

M.A. Economics: University of Maryland - College Park

B.A. Economics: University of Utah, *Magna cum Laude*

PROFESSIONAL EXPERIENCE:

Director – Economics and Policy: John Staurulakis, Incorporated, Seabrook Maryland (1998-present).

Responsible for the development of economic studies required by various regulatory authorities. Also responsible for overall policy development for JSI in arena of the federal and state telecommunications policy. I work in the development of position papers on various policy topics. I participate in the policy development of national originations and associations. Present policy positions through workshops and testimony. Currently serve as economic advisor for the Puerto Rico Telecommunications Regulatory Board.

Senior Economist: John Staurulakis Incorporated, Seabrook, Maryland (1995-1998).

Duties included the preparation of economic cost studies and feasibility studies for telecommunications clients. Responsible for the development of Internet services for rural telecommunications carriers.

PRESENTATION EXPERIENCE:

Participation in presentations and national association meetings and institutes are not listed individually. Approximately five presentations performed annually.

INDUSTRY COMMITTEE EXPERIENCE:

Former Member of OPASTCO Separations and Access Committee
Former Member of OPASTCO Universal Service Committee
Former Member of RUPRI Telecommunications Policy Committee

HEARING AND TESTIMONY EXPERIENCE:

[SEE ATTACHED TABLE]

Date	Case Number	Case Name	Party
2 Oct 1996	U-22022	Proceeding investigating BellSouth Cost Study	Small Company Committee
20 Oct 1997	3650-MA-100, 5845-MA-100	On behalf of Mid-Plains, Inc. Before the Arbitration Hearing for the Public Service Commission of Wisconsin	Mid-Plains, Inc.
30 Oct 1997	Docket No. 5713, Phase II, Module 2	NYNEX TELRIC Study	Vermont Independent Telephone Members
17 Nov 1997	97-239-C	Development of statewide universal service disbursements	South Carolina Telephone Coalition
1997	Not docketed	Mediation Proceeding involving transport and termination rates	Wood County Telephone Company
1998		Arbitration Proceeding determining Horry Wholesale Discount	Horry Telephone Cooperative, Inc.
Nov 1998	Mediation	Arbitration proceeding determining Vista-United Telecommunications Wholesale Discount	Vista-United
Nov 1998	Case 98-C-1249	Proceeding to Consider Petition of Warwick Valley Telephone Company for Mediation of an Interconnection Agreement with Citizens Telecommunications Company of New York, Inc. and any Resulting Interconnection Agreement.	Warwick Valley Telephone Company
10 Mar 1999	PU-1564-99-17	Proceeding to Consider Complaint of Western Wireless Corporation d/b/a Cellular One vs. Consolidated Telephone Cooperative, Inc.	Consolidated Telephone Cooperative, Inc.
27 May 1999	Case No U-11996	In the matter of the application of Hiawatha Telephone Company for approval of a total service long run incremental cost study	Hiawatha Telephone Company
Sept 1999	A1-99-0006	Western Wireless Corporation vs. Consolidated Telephone Cooperative, Inc.	Consolidated
31 Jan 2000	PU-1564-98-428	Proceeding to Consider the Western Wireless Corporation Designated Eligible Carrier Application	15 rural LECs located in North Dakota
23 Jun 2000	PUC Docket No. 22289	Application of WWC Texas RSA Limited Partnership for Designation as an Eligible Telecommunications carrier pursuant to 47 U.S.C. § 214(E) and P.U.C. Subst. R. § 26.418, as an Eligible Telecommunications Provider Pursuant to 47 U.S.C. § 214(E) and P.U.C. Subst. R. § 26.417	Texas Telephone Association and Texas statewide Telephone Cooperative, Inc.
15 Feb 2002	DT 00-223	Investigation into whether certain calls are local	8 Independent Telephone Companies of New

Date	Case Number	Case Name	Party
			Hampshire
Sep 2002	Case No. PU-2077-02-308	Petition of WWC Holding Co., Inc. For Arbitration Under the Telecommunications Act of 1996.	22 Independent Telephone Companies of North Dakota
4 Dec 2002	Case No. PU-2065-02-465	In the Matter of the Petition of Level 3 Communications LLC, Interconnection Arbitration Application	SRT Communications, Inc.
11 Sep 2003	Docket No. 2003-151-C	Application of ALLTEL Communications, Inc. for Designation as an Eligible Telecommunications Carrier Pursuant to Section 214(e)(2) of the Communications Act of 1934	South Carolina Telephone Coalition
Sep 2003	TC02-176	Petition for Arbitration on behalf of WWC	30 Independent Telephone Companies in South Dakota
2003	CV 01-163-BLG-RFC	Mid-Rivers Telephone Cooperative, Inc. v. Qwest Corporation	Mid-Rivers Telephone
15 Apr 2004	04A-018T	In the Matter of WWC Holding Co, Inc. Application for Designation as an Eligible Telecommunications Carrier and Redefinition of Rural Telephone Company Service Area Requirement	Colorado Telecommunications Association and Rye Telephone Company
15 Jun 2004	Docket No. 6934	Petition of RCC Atlantic, Inc. for designation as an Eligible Telecommunications Carrier in areas served by rural telephone companies under the Telecommunications Act of 1996	9 Independent Telephone Companies of Vermont
May 2005		In the Matter of the Petition of MCI, Interconnection Arbitration Application	4 Independent Telephone Companies in South Carolina
Sep 2005	Docket No. 2005-188-C	In the Matter of the Petition of MCI, Interconnection Arbitration Application	Horry Telephone Cooperative, Inc
14 Dec 2005	Docket No. 05-2302-1	Petition for increasing rates and application for Utah universal service support	Carbon/Emery Telecom
13 Jan 2006	Docket No. 2005-219-C	In the Matter of the Petition of Budget Phone ETC Designation	South Carolina Coalition
23 Jan 2006	PU-05-451	Midcontinent Communications v. North Dakota Telephone Company	North Dakota Telephone Company
10 Jul 2006	PU-05-451	Midcontinent Communications v. North Dakota Telephone Company	North Dakota Telephone Company
20 Jul 2006	Docket Nos. 2006-137-C,	Petition of Charter Fiberlink for Arbitration of Interconnection Agreement	4 Independent Telephone

Date	Case Number	Case Name	Party
	138-C, 139-C and 142-C		Companies in South Carolina
23 Jun 2006	Case No. U-14782	In the Matter to examine total service long run incremental costs of Upper Peninsula Telephone Company	Independent Telephone Company in Michigan
7 Feb 2007	Case No. U-15035 14781	In the Matter to examine total service long run incremental costs of Chippewa County Telephone, Hiawatha Telephone Company, Midway Telephone Company, and Ontonagon Telephone Company	4 Independent Telephone Companies in Michigan
27 Apr 2007	06-00228	Tennessee Rural Independent Coalition Petition for Suspension and Modification	Tennessee Rural Independent Coalition
16 Jul 2007	07-2476-01	Application of Bresnan Broadband of Utah for a CPCN	URTA – Utah Rural Telecom Assoc.
16 Apr 2008	PU-08-58	Midcontinent Communications / BEK Communications Cooperative Arbitration Applications	BEK
21 Jul 2008	PU-08-97	Midcontinent Communications / Consolidated Telecom – Arbitration Petition	Consolidated Telecom
3 Sep 2008	Docket No. 5-MA-147	Charter Fiberlink, LLC Petition for Arbitration of Interconnection Rates, Terms, Conditions, and Related Arrangements with Wood County Telephone Company Pursuant to 47 U.S.C. § 252(b)	Wood County
2009		ETC (Sunman) Gross Receipts Tax testimony in Indiana Tax Court	ETC
Dec 2008	2008-325-C	Application to amend Time Warner CPCN (South Carolina)	Rural incumbent carriers
Jan 2009	08-2476-02	Petition for interconnection of essential facilities (Utah)	Strata f/d/b/a UBTA-UBET Communications, Inc
2009	Various	TDS v. Time Warner – duty to negotiate	TDS
3 Jul 2009	2007-00004	Brandenburg et al. v Windstream Kentucky East, Inc.	Brandenburg et al.
2009-2010	Docket No. 2009-41 through 2009-44	Oxford, Oxford West, Lincolnville, Tidewater Maine Rural Exemption Proceeding	Oxford, Oxford West, Lincolnville, Tidewater
October 2009	Docket No. DT 09-044	Regulatory Status of IP regulation	New Hampshire Telecom Association
Dec 1, 2009	Docket No. RT-	In the Matter of the Review and Possible Revision of Arizona	Arizona Local Exchange Carriers

Date	Case Number	Case Name	Party
	00000H-97-0137	Universal Service Fund Rules, Article 12 of the Arizona Administrative Code.	Incumbent Carrier Association
February 12, 2010	DOCKET NO. 08-2469-01	In the Matter of the Petition of All American Telephone Co., Inc. for a Nunc Pro Tunc Amendment of its Certificate of Authority to Operate as a Competitive Local Exchange Carrier Within the State of Utah	URTA
September, 30, 2010	Docket No. 10-049-16	In the Matter of the Joint Application of Qwest Communications International, Inc. and CenturyTel, Inc. for Approval of Indirect Transfer of Control of Qwest Corporation, Qwest Communications Company, LLC, and Qwest LD Corporation	URTA
Dec 2010	DT 10-183	Petition by rural Telcos regarding CLEC registrations (New Hampshire)	Rural incumbent carriers
2011	2011-243-C	Petition for Arbitration of Interconnection Agreement between Time Warner Cable Information Services and Four rural incumbent carriers (South Carolina)	Rural incumbent carriers
2012	Docket No. 7798	Section 251(f)(2) Suspension and/or Modification for Waitsfield Champlain Valley Telephone Company and Comcast (Vermont)	WCVT
2012	Docket No. 2012-133 to 2012-136	TIME WARNER CABLE INFOMRATION SERVICES (MAINE) LLC, Request for Arbitration of Interconnection Agreement Between Time Warner Cable Information Services (Maine) LLC & Four rural carriers	Rural telephone carriers
2012	Docket No. 2012-218 to 221	Petition for Section 251(f)(2) Suspension and/or Modification for Four Maine RLECs	Rural telephone carriers
2013	Docket No. FC 1090	Investigation into the Reliability of Verizon Washington, DC's Telecommunications Infrastructure	Office of Public Counsel
2013	I.09-12-016	TracFone Wireless, Inc. (U-4321-C) to collect and remit public purpose program surcharges and user fees on revenue from its sale of intrastate telephone service to California consumers	TracFone
2014		Petition for State of Maine Universal Service Funds	FairPoint, ME

Date	Case Number	Case Name	Party
5 Sept 2015	15-2302-01	Carbon/Emery Application for an increase in Utah Universal Service Fund Support	Carbon/Emery and Utah Rural Telecom Association
18 May 2016	PDS 14-01	PACIFIC DATA SYSTEM INC.'S PETITION FOR RBRITRATION OF – TELRIC MODEL INTERCONNECTION AGREEMENT	TeleGuam Holdings, LLC

Section 251 Arbitration proceedings in Puerto Rico			
1997-2001	Various	Various	Advisor to the Puerto Rico Telecommunications Regulatory Board (Board Advisor)
2001	JRT-2001-AR-0002	WorldNet Telecommunications, Inc. and Puerto Rico Telephone Company, Inc. (PRTC)	Board Advisor
2002	JRT-2002-AR-0001	Newcom, Comm Wireless Services d/b/a MoviStar and PRTC	Board Advisor
2002	JRT-2002-AR-0002	Centennial PR and PRTC	Board Advisor
2003	JRT-2003-AR-0001	WorldNet Telecommunications, Inc. and Puerto Rico Telephone Company, Inc. (PRTC)	Board Advisor
2005	JRT-2005-AR-0001	Centennial PR and PRTC	Board Advisor
2006	JRT-2006-AR-0001	TLD and PRTC	Board Advisor
2007	JRT-2007-AR-0001	WorldNet Telecommunications, Inc. and Puerto Rico Telephone Company, Inc. (PRTC)	Board Advisor
2007	JRT-2007-AR-0002	Sprint and (PRTC)	Board Advisor
2008	JRT-2008-AR-0001	Centennial PR and PRTC	Board Advisor
2010	JRT-2010-AR-0001	WorldNet Telecommunications, Inc and PRTC	Board Advisor
2012	JRT-2012-AR-0001	Liberty Cable, Inc and PRTC	Board Advisor

DDM-2

South Carolina Telephone Coalition Member Companies

Bluffton Telephone Company, Inc.
Chesnee Telephone Company
Chester Telephone Company, d/b/a TruVista
Comporium, Inc. (f/k/a Rock Hill Telephone Company)
Farmers Telephone Cooperative, Inc.
Ft. Mill Telephone Company, d/b/a Comporium
Hargray Telephone Company, Inc.
Home Telephone ILEC, LLC d/b/a Home Telecom
Horry Telephone Cooperative, Inc.
Lancaster Telephone Company, d/b/a Comporium
Lockhart Telephone Company, d/b/a TruVista
McClellanville Telephone Company (TDS)
Norway Telephone Company (TDS)
Palmetto Rural Telephone Cooperative, Inc.
Piedmont Rural Telephone Cooperative, Inc.
PBT Telecom, d/b/a Comporium
Ridgeway Telephone Company, d/b/a TruVista
Sandhill Telephone Cooperative, Inc.
St. Stephen Telephone Company (TDS)
West Carolina Rural Telephone Cooperative, Inc.
Williston Telephone Company (TDS)

DDM-3

6

VoIP AND UNIFIED COMMUNICATIONS DEFINE THE FUTURE

Knowing some history of telephony and understanding the new technologies should prepare you to face your specific business problems. The intent is to help evaluate, select, and deploy future communications systems for at least the next decade.

6.1 VOICE AS BEFORE, WITH ADDITIONS

The goal of UC is to improve communications, which means the mark of success will be increased usage per person. What new functions will a migration add? What effect will it have on network traffic? The direction is up, but where will you start?

Before that increase hits, you should calculate if the IP network is able to absorb voice and UC in addition to data. A PBX will provide some demand information; phone bills are another source.

When planning basic capacity, compare what you know to the assumptions that vendors make when recommending the size and number of servers. Some vendors openly state their basis for a calculation, for example, in terms of the

number of emails, phone calls, and IM messages each person makes in a day or hour. If you don't validate the assumptions against your own data, any prediction will be a wild guess and probably wrong.

The same applies to assumptions about costs when calculating a potential return on investment. More on that below.

RFC 5359, *Session Initiation Protocol Service Examples*, collects best practices for legacy services using SIP "methods" and protocols. Watch for similar documents to emerge in the future, not only from the IETF but also from vendors, associations, and industry forums.

6.2 LEGACY SERVICES TO KEEP AND IMPROVE WITH VoIP

You can still get directory service, but 411 isn't free any more. Cellular carriers will dial the number they look up for you, either for no additional charge or for a fee. The correct time is available in most areas for a local call (but is being discontinued by major carriers). The author's audible time reading was within 2 seconds of the computer display controlled by the Network Time Protocol (NTP). These you might want to keep.

Astrology readings, gambling hints, and recorded financial advice? Might not need to preserve them. With digital controls, it is possible to block connections in the same way that parental controls can block websites. SIP phone numbers and web addresses are all URLs or URIs.

The point here is that even analog phones provide more than a voice connection. Unified Communications will provide far more functionality, some new and some carried over. You will have to decide which services your system provides, with a little help from your friends in local, state, and federal government.

Once completely uncontrolled, VoIP has become too important to leave to end users and peer-to-peer desktop applications. The call volume carried on IP networks exceeded the volume on TDM networks for international calling around 2010 and continues to increase. VoIP service providers such as Skype and Vonage avoided regulation as telephone carriers for years but became too large to ignore by taxing authorities. In 2011 the requirements for E911 location reporting and certain taxes applied to almost everyone providing voice service. Certainly enterprise users need to comply with E911 laws, which are becoming more demanding in more states each year.

On the positive side, a VoIP provider classified as a local exchange carrier (LEC) gets to control public (E.164) telephone numbers, direct inward dial (DID) numbers, and can assign them to its customers. A LEC also is entitled to handle the phone number of a customer who chooses to take that number from another carrier—though this ability to port local numbers is advancing more slowly.

6.2.1 Flexible Call Routing and 800 Numbers

What exactly does a phone number stand for? How does the network find that phone? The introduction of packet switching for voice not only unifies communications but also changes the answers to those questions.

Legacy PSTN switches present a separate hardware interface for each analog phone or trunk connected to the switch. Some of these ports are in the central office, some at remote terminals. Originally the identification of that physical port, or connector, was the directory number (DN), the telephone number in the phone book that you dialed to reach that phone. This meant that a phone number corresponded to a specific local loop that extended to a specific location from a specific switch. The phone number represented the region (area code), the central office (the exchange), and the phone line (the last four digits).

To encourage people to adopt the telephone, the Bell System business model imposed billing on the caller—there was no incremental charge to receive calls (beyond the monthly connection fee). If a business wanted to save its customers toll charges, to encourage them to call from a distance, it would obtain a local number in another area served by a different switch. A business had to lease a private line from the switch that controlled the desired number the business wanted to make available to its customers. That foreign exchange (FX) line passed through the business's own telephone central office to a local loop. FX let a business have local numbers outside the service area of its own CO. The monthly FX line charge was by the mile, so it was expensive and economically practical only over a small region or metro area.

From early on in the Bell System it was possible to call and “reverse the charges” (have the called party pay), but that required the intervention of a live operator and acceptance by a person who answered the call. The process was a bother for the caller and also very expensive for the callee. But business wanted to encourage customers to call and were willing to pay. To fit the billing system, businesses that paid for incoming calls received virtual phone numbers, the 800 numbers, which were handled automatically as reversed charges. 800 numbers changed what the telephone number could represent. Now it could be any DN, anywhere in the country.

To map an 800 number to a DN that the switches understand requires a lookup in a database (Figure 3.1). This was not difficult when only 800 numbers required a translation. A software change enabled the switch to find a DN when it received an 800 call, then use the DN to make the connection.

VoIP numbers and SIP addresses likewise represent any location. The improvement is that the new addressing reaches anywhere in the world.

6.2.2 Call on Hold

Typically applied to voice sessions, the “hold” function cuts off the audio transmission while keeping the connection. The original hold button activated a mechanical switch to isolate the hand set. In VoIP, the initiator of the hold changes the state of the media session while keeping the control session open.

In SIP, a re-INV message from either end carries an SDP body in which one of the attributes (directionality) for the session is changed to:

- a = inactive if the initiator does not provide music on hold.
- a = sendonly if the end placing the hold will provide MOH.

6.2.3 Call Transfer

A UAS that accepts a call (UA = B) can transfer the caller (UA = A) to another number (UA = C) by using INV and re-INV messages. The first session/dialog (A–B, based on SD-1 and SD-2) is put on hold (see above). Then:

- UA-B sends an INV with no session description to the transfer target (C) to create a second dialog.
- The target C responds with an OK 200 message containing an SDP offer (SD-3), which contains its contact information and capabilities.
- UA-B puts SD-3 into a re-INV message on the first dialog to UA-A, who now has C's information.
- UA-A sends an OK on the first dialog, with a session description SD-4.
- UA-B puts SD-4 into the ACK to C on dialog 2.
- UA-B ACK's A.

If the user at UA-B hangs up before C answers, the system should complete a blind transfer. If B talks to C and then hangs up, it is a supervised or attended transfer, just like the old days.

6.2.4 Call Forwarding

A UA proxy may handle call forwarding for a phone registered with it. The server may apply a filter, forwarding only certain calling numbers or calls at certain times of day, to voice mail or another destination. The end user may also tell the proxy to forward calls to a temporary location by changing the primary registration.

To effect a forward, the proxy issues an INV message to the new target phone (or terminal or server; fax calls can be forwarded too, e.g., on detecting a fax modem tone), creating a second dialog. The message includes the SDP information from the caller and the caller's contact information. It is not required for responses to come back through the proxy if it is stateless.

6.2.5 Audio Conferencing

Conference call service used to be highly profitable. The audio bridge was large and expensive, not something most enterprises wanted to own. Governments lacked capital budgets, but they could pay by the minute, per participant line, to discuss projects. Travel savings justified the cost of conferencing.

More powerful computer chips and cheaper memory brought down the cost of a bridge. Even early digital PBXs offered bridging for a handful (or two) of participants as part of the feature set.

Digital signal processors (DSPs) brought the cost down further, enabling bridges to conference thousands of lines on a call. These large devices allowed rural local exchange carriers (LECs) to offer free conferencing service. Participants called, mostly from a distance. The LEC collected call termination fees paid by interexchange carriers (IXCs, long-distance network providers). Conference participants pay for the call to the free bridge. For a fee, typically paid by the host company, the bridge owner will let participants call on a toll-free number.

In the improved UC arena, conferencing has become standard at no additional cost in both hardware and software products. It may reside in a media gateway, drawing on DSPs in the hardware. Or a software bridge may run on a call control server or on a dedicated device such as the H.323 Multipoint Control Unit (MCU).

Distinguishing differences among UC conference systems that can affect your business are:

- Number of ad hoc conferences at one time.
- Ability to schedule conferences to reserve resources for large calls.
- Total number of lines simultaneously in one or more conferences.
- Controls available to a conference moderator:
 1. Mute all lines or individuals.
 2. Choose video feed to distribute; from participants, speaker, or designated source.
- Authorization or authentication methods to control admission to a conference for privacy
- Facility to distribute documents or other files, via on-demand download or pushed by the host

6.2.6 Video Conferencing

Video conferencing scored high among end users when asked about the features they most desired in a Unified Communications environment. Video to date largely has been either one-way broadcast to a large audience (announcements, coverage of live events, speeches, etc.) or a meshed exchange of images among a relatively small number of users or sites. How you want to use video leads to another item on the previous list:

- Video procedures and image switching/combining during a conference or broadcast.

Three- to five-way video conferences allow all participants to view each other with at least a quarter of the screen per site in a static configuration.

Among larger groups, the best results come from switching the main feed to whichever site is producing the audio at the moment. The more sophisticated video bridges or Multipoint Control Units (MCUs) can tie the video feed to the current audio source.

Telepresence goes further by applying stereo sound imaging to place a speaker's voice with the image.

6.2.7 Local Number Portability

A later and much larger version of the 800 number data base now includes more area codes for toll-free numbers (866, 877, and additional free-call exchanges as needed). The same technology potentially applies to every phone number because of the US national policy called Local Number Portability (LNP). The Federal Communications Commission requires most carriers to support LNP by giving up or receiving the number of a customer who wants to change service providers but keep the phone number.

It's called local portability because the original scheme restricted the change to land lines in the same area code. Later cellular numbers were added. However, with the prevalence of unlimited long-distance calling, the importance of having local numbers or 800 numbers has declined.

Nevertheless, if you've grown attached to your phone number, or spent heavily to advertise it, you probably want to keep that number when moving from TDM service to VoIP. The FCC says you can do that. Routing calls from the PSTN to a VoIP system uses the same database that controls 800 calls. Either the called or calling party's carrier may operate a gateway that receives these calls and places them on an IP network. You may retain a gateway that receives calls on traditional TDM trunks, until replaced by SIP trunks.

6.2.8 Direct Inward Dialing, Dialed Number Indication

In the PSTN, dialed number indication (DNI) allows a call through a PBX (private branch exchange, the enterprise phone switch) to ring directly on an extension without requiring action by an attendant. Telcos call this feature direct inward dialing (DID) or dialed number indication service (DNIS). DNI lets the PBX route a call to a specific phone inside an organization.

DID depends on assigning a public (E.164) directory number (DN) to a phone. LECs charge a small monthly fee for DID numbers, which are assigned in blocks that may not be related to the primary phone number. Phones on a PBX that lack a DID number can call out by requesting a trunk (dial 9), but incoming calls must pass through the attendant. It is up to the customer (you) to have DID numbers for all extensions in addition to a general number that usually rings at an attendant station.

ISDN digital trunks from the CO always carry DNI in the call request in the format of the Q.931 packet signaling protocol. A PBX with a PRI (an ISDN trunk) interface understands the message. A media gateway that receives a

PSTN call extracts all the information from the Q.931 message and passes it in a SIP message to the call control server or media gateway controller (MGC).

If the central office switch sets up a DID call to a PBX on an analog trunk, the CO switch treats the PBX like another PSTN switch on a tandem trunk. The CO passes DNI to the PBX (or the MGW) during the 4 seconds after the first full ring (which is 2 seconds of 20 Hz a.c. imposed on the line to announce a call). COs use one of several signaling methods to send DNI on POTS lines:

- DTMF digits, TouchTone sounds.
- An asynchronous data protocol message of ASCII characters at 1200 bits/s. Frequency-shift keying, a V.23 modem signal, sends tones of 1200 Hz (1 or mark) and 2200 Hz (0 or space) to represent binary digits. There are detailed signal and timing requirements. The message includes the date, time, and a checksum to confirm accurate delivery.
- Multifrequency tone pulses, where each pair of six audible frequencies (odd 100s from 900 to 2100 Hz) represents a dialed digit (or a function, e.g., coin return). This older mode is as common as pay phones—not very.

Media gateways may be configured to receive DNI information from the PSTN and will pass the called number to the UA server that routes connections. The DNIS method on the MGW must match what the central office switch uses. In addition the call control signaling must also match. MGWs, in general, support FXO, FXS, loop or ground start, and possibly wink start and reverse battery. The carrier will supply this information about legacy trunks.

Reviewing data sheets for MGWs, it appears that not all will accept in-band DNI information (DTMF, MF). Those with only digital interfaces (T-1, E-1, ISDN) seem to prefer ISDN D-Channel signaling over channel-associated signaling (robbed-bit signaling) to receive DNI.

SIP trunking takes a view of DID and DNI similar to ISDN. All SIP calls are DID, targeted to a specific recipient. A SIP address, a URI of the form SIP: name@domain.tld or similar, identifies the called party so the recipient UAS always has a specific identity to locate (or group of phones). The attendant is just another specific DID address.

6.2.9 Call/Message Waiting

Media gateways in front of a PBX can recognize the signals for a waiting message:

- 100 V applied to the loop.
- Stutter dial tone.
- Proprietary signals to digital phones.

A SIP message to the IP phone turns on the light or triggers the display. For phones without a display the system can create a “stutter” dial tone.

Call-waiting tones and messages from a PBX are converted to a SIP Notify message. The recipient can see the information on the phone's display if it has one. If the target phone is on an FXS port, the gateway generates the message waiting indication (MWI), an audible tone heard only by the one party.

6.2.10 Call Recording

One reason to focus voice traffic at one point in the network is to record conversations—for “quality control or training purposes” as well as documenting financial transactions such as brokerage orders.

Several vendors offer customer premises equipment (CPE) for recording. Having your own system allows you to tag calls with date, time, employee number, customer phone number, internal extension number, and other data to facilitate retrieval on demand. Carriers and VoIP service providers who host call control may also offer call recording.

A choice between a service bureau and your own equipment will be influenced by cost but also by how often recordings need to be retrieved and the relative ease and speed to recover a specific conversation. Concerns about confidentiality may point to on-premises recording.

Many routers, media gateways, and session border controllers have the ability to duplicate media streams and send the copy to a recorder. There the packets are saved to disk. As with most collections of information, the disks will fill. Anticipate running out of disk storage with a policy to discard earlier recordings or to back them up to another device or medium.

An example is the Alcatel-Lucent RECORD suite of software. It will record not only the voice element but screen images of agents. It works with legacy phones and, through a MGW, IP phones.

6.2.11 Emergency Calling (E911)

When 911 appeared, the phone company programmed a translation into the CO switch for your phone. Dialing 911 sent the call to a local public service answering point (PSAP). That is, 911 was translated into a DN for a line to the PSAP, similar to the handling of calls to 800 numbers. PSAP lines are configured for centralized automatic message accounting (CAMA) to deliver the calling phone number, the calling line identification (CLID). An ISDN PRI trunk also will do the job in a different format. The idea was that the PSAP could call back if necessary.

At first the choice of PSAP reflected the physical location of the telephone line demark for the CLID. That is, the address where the telco terminated its local loop on inside wiring was reported to the PSAP. This is all the detail that a LEC is certain to know in every case. For a single-family home, the location is clear. A reverse directory lookup gives the address.

In a multitenant building, commercial or residential, the demark could be inside any office or apartment at the address. It's still reasonably specific

because the line carries a customer account name or apartment number. But things got messy.

The CLID itself no longer indicates an address without ambiguity when:

- Campus environments terminate phone lines in a PBX room, but the extensions cover many separate buildings, often far apart.
- Landlords operate phone services in large buildings, hiding customer names and exact locations behind a shared demark.
- Large corporations occupy dozens of floors in a skyscraper, all with the same address and general phone number.

People died because ambulances went to the main building in response to a call from another location that passed through the one PBX serving both locations.

As PSAPs evolved, the CLID became less certain to be specific. For example, an extension that lacks a DID number shows the general line as the CLID. A call to that rings at the attendant station, possibly in a different building. The need was to show a CLID that represented the location of the phone rather than the demark. PBX vendors modified their switches to provide a more flexible CLID, but much of the older equipment was not highly capable for this.

The CLID became an index to Public Service—Automatic Location Information (PS-ALI). This is a database that holds actual location information on a location information server (LIS). The local exchange carrier (LEC) or its contractor maintains the LIS for each region.

For each call received, the PSAP uses the CLID to retrieve the caller's address. The screen pop requires a dip into the ALI database. At a minimum, the ALI contains a street address. A perceived need for more specific location information grew into legislation in several states, and federal initiatives for Enhanced 911. E911 makes space for at least an additional 20 characters that can further identify the location. The enhanced database record can be more specific by adding a building number, floor quadrant, office number, or a combination of pointers to deliver a more specific emergency response location (ERL).

An ERL should represent no more than a zone on a specific floor in a building. In the most demanding jurisdictions a call to 911 must generate an ERL to identify the location of the calling phone within 100 feet. Other jurisdictions' requirements can be less strict. Almost all state laws have several points in common:

- 911 calls must not be blocked, either deliberately (to prevent prank calls) or accidentally (by not planning for them in a new phone system).
- ANI (caller ID, CLID) must be sent with a 911 call.
- Specific location data complying with local law is required; more is better, send as much as possible, including the calling extension number.

The oldest PSAPs remain tied to the ALI and CAMA trunks. The large number of jurisdictions with public service answering points (PSAPs) and a push to update them to handle more modes of communications means that the state of 911 communications is in flux. Within a decade expect development of “Emergency Context Resolution with Internet Technology” (ECRIT) to change the PSAP’s connections. Work proceeds in the IETF, with several drafts undergoing work. Eventually, the Emergency Services IP Network (NSInet) in each region will tie together the PSAP(s), first responders, medical facilities, and emergency offices.

Large organizations that have a security staff will want notification of 911 calls in progress sent to the watch desk, with the location. Security staff can speed medical responses by alerting check-in points to admit ambulance crews, calling and holding elevators, and so forth.

Next Generation E911 will enable a PSAP to accept text messages, SMS messages, GPS location information from a caller’s cell phone, and VoIP calls via SIP with the location information in a newly defined field of the INVITE message. A client-server architecture will be more adaptable to future technologies.

A more flexible stand-in for the CLID is the emergency location identification number (ELIN). This continues to be the phone number that the PSAP can use to call back, and the index to the location database, but it no longer need refer to the caller’s specific phone. The ELIN may be a CLID, but it can also refer to a location zone that includes many phones, one of which the PSAP can reach by DID. An updated PBX reports the caller ID on a 911 call as the ELIN, not the main number or that extension’s DID number. A good ELIN might ring at the desk of an employee designated as assistant emergency coordinator for the zone.

An organization obligated to provide detailed location information for the ALI has several ways to conform. The choices are roughly the same for a legacy PBX or a VoIP system.

- Install POTS lines at each location, that is, in each location zone. Route 911 calls to them, and let the LEC maintain the ALI. Changes can be slow to make. The LEC updates records once per day typically.
- Purchase a “gateway account” from the LEC to gain access to the Public Safety–ALI database. Each organization manages the records for its own lines and numbers. The gateway device costs several thousand dollars. The charge to store records is about \$1 per year each. Updates take effect quickly when delivered to the gateway.
- Hire a contractor to process orders for phone moves/adds/changes into updates to ALI records. This process requires the customer to buy an ALI gateway, on-site server software, and the support service.
- Adopt E911 as a service. Small and medium businesses (SMBs) can put everything on the contractor, “in the cloud.” Larger firms may need to stand up one or more servers to track all of its phone locations and help handle 911 calls.

Dealing directly with the PS-ALI database works best for firms with no more than a few hundred phones and not much movement of those phones. Larger firms may want the help of specialized server software, such as a 911 manager, a LIS, and voice positioning center (VPC). Firms with wide geographic presence or many remote offices may want a contractor that tracks the more than 5000 PSAP areas by city, county, and so forth. Political jurisdictions often share a PSAP so the mapping of PSAP coverage requires detailed local knowledge down to the street address.

Alternatively, an enterprise may engage a contractor to build and maintain a database of phone locations and zones. The “full-service” E911 contractor may process all 911 calls for an enterprise. Commercial services that provide E911 location information with calls to a PSAP offer both a local database of locations, within a campus, for example, and an interface to the public LIS. Either may satisfy the requirements for E911 calls from a corporate office, depending on the extent of the private network and the costs to build and maintain lists in the two environments.

The architecture for one vendor involves:

- A 911 server on customer premises, the location information server (LIS), that maintains a database of phone locations both inside and outside the private network.
- An IP-based call control server to forward 911 calls to the contractor’s data center.
- A dedicated IP connection between the enterprise call control server and the contractor’s voice positioning center (VPC), that matches locations to the PSAPs that cover them.
- A SIP-based IP channel to carry 911 calls from the enterprise call server to the contractor.
- Notification to the enterprise’s security office.

All 911 calls pass through the enterprise call server, which picks up the phone’s location from its own VPC. The call becomes a SIP INVITE message (containing the calling number and its location) to the contractor’s softswitch. That switch uses the location to complete the call:

- Pick the appropriate PSAP and its regional PS-ALI database.
- Create an ALI record for the current location of the phone and an ELIN.
- Insert that ALI record in the regional ALI database.
- Find the telephone routing number for the PSAP from the master list.
- Forward the 911 call to an Emergency Services Gateway (ESGW) in the PSTN using the national Transport Number (TN). The 911 transport network then delivers the call to the selected PSAP.

An E911 feature is built into the firmware of routers intended for small or branch office locations that normally rely on the IP WAN for telephone service.

The 911 calls don't go to the VoIP call controller but are routed immediately to a local POTS line so that they reach the appropriate PSAP. For a small site there's no effort on the customer's part to maintain the PS-ALI database.

If the call passes through a media gateway to a circuit switched line such as a PRI, the gateway inserts the ELIN as the calling phone number in the signaling message. Calls other than to 911 carry the phone's DID number for the standard operation of caller ID.

6.2.12 Tracking IP Phone Locations for E911

VoIP phones can plug in anywhere. A user can move a phone to a different floor, take it home, or pack it on a business trip. When the phone or user authenticates to the call server, it has the same phone number but a different IP address. When an enterprise tracks migrated phones, it will also update the public service ALI database (the LIS) so that the PSAP finds the location for a phone number. In a future IP environment, the mobile or nomadic device will discover its own location, for example, via GPS or a location service of a carrier.

One common way to resolve the location of an IP phone on the enterprise network relies on its IP address. The network operator assigns IP addresses in the enterprise's Dynamic Host Configuration Protocol (DHCP) server so that every subnet address range falls into a known and controlled range of physical cabling within a defined floor area no larger than that legally required for an ELI. The location radius varies widely by political jurisdiction.

Configuration of the DHCP server restricts IP address assignments available to each zone. Any device that plugs into the network in that zone will receive an IP address, via DHCP, from the IP range assigned to that zone. When the phone registers with the call server, an important parameter is the phone's IP address. A 911 management server receives notifications of new registrations from the call processor, including the IP address in each new registration. This interface may require some system integration, often performed by the two software vendors in a partnership that enhances the value of both products.

The 911 server examines the database of IP address assignments to identify the location zone, and tells the call server the telephone number to use as the emergency location identification number (ELIN) when that phone calls 911. The call processor handles 911 calls by associating the calling phone with the corresponding ELIN, not the phone's DID number or DN. Thus the PSAP dips into the ALI database for the ELIN, which returns the correct location of the phone's zone.

Because the PSAP needs the caller ID, the SIP Forum's SIP Connect recommendation for carrier trunking requires behavior that conflicts with privacy requests. When a SIP-PBX places an emergency call it must not withhold caller ID even if the caller asserts a right to privacy on that information by including a P-Asserted-Identity header.

TABLE 6.1 Two methods to identify an internal location of an IP device

	IP Address/vLAN	Switch Port
IP address assignment	By DHCP	By DHCP
Extent of smallest zone	Subnet or vLAN	Cable drop or switch port
How 911 server resolves location	Configured table of IP's (same as in DHCP server)	SNMP search for switch port with registered IP and MAC addresses
How 911 server selects ELIN	Table matches IP address to ELIN zone	Table matches switch port to ELIN zone

Another way to locate an IP phone depends on mapping physical switch ports that terminate LAN drops to phones. Each port or group of ports is assigned to a location zone in the 911 management server. When an IP phone registers with the call control server, it notifies the 911 server, which searches the network for that IP address using Simple Network Management Protocol (SNMP). When found, the port is identified and with it the location zone. This approach resembles the earliest assumption that a number represents a location.

Accuracy in placing IP phones into zones depends on keeping the list of assigned IP addresses or switch ports up to date. Precise recording of the as-built configuration tends to get neglected. An outside contractor that specializes in network audits may prove more reliable. Automated discovery software is often a help in IP address management.

A record associating each ELIN with a location zone must be added to the SP-ALI database. This process may be manual, but automation in the 911 server, through a gateway, speeds the work for large networks and increases accuracy.

When the enterprise's VoIP call processor sees a 911 call, the ELIN is inserted in the call request or INVITE message as the caller's phone number.

Nomadic users of IP softphones pose a particular problem. They will acquire IP addresses from DHCP servers in hotels and airports that may not have a strictly defined geographical area that maps cleanly to a PSAP. It is possible to include these IP phones if the user will input a location when on trips.

There's at least one application for that, which runs on the same notebook computer running the softphone and links to the 911 management server. This application intercepts the registration process to interpose an additional step. Before the call control server will accept a remote softphone, the user must enter an address and any additional location details into the 911 server. That server then validates the address in the national Master Street Address Guide (MSAG) before allowing the phone to operate.

Upon verifying the location, the 911 server creates an ALI record and stores it in a temporary ALI. It is only when that phone makes a 911 call that the temporary ALI record is inserted into the proper regional SP-ALI database where the PSAP can retrieve the location.

Note that the phones do not need to be IP based. The location database for legacy PBXs has been populated manually but automated procedures are also used. For example, the human resources record typically includes a phone number and a workstation address for delivering postal mail. Those data can fulfill the E911 requirement.

A PSAP may convert to VoIP, on the way to UC, before the NG911 network becomes available. The site would insert a special purpose MGW's function to terminate the CAMA trunks from the CO switch (Figure 6.1). The gateway converts a call from the PSTN in the legacy format to a SIP INVITE addressed to the PSAP's proxy.

The E911 call that originates on the PSTN follows the legacy procedure as far as routing by the serving office to a 911 tandem switch. The call appears on a trunk to the PSAP, shown in the example as a loop-start analog trunk.

The E911 tandem switch seizes the line, in this case by closing loop and drawing current from the gateway. The FXS device acknowledges the line seizure with a wink signal (250 ms of reverse battery). Seeing the wink, the tandem sends the MF "Spill," the digits with ANI. The gateway creates a SIP INVITE message to the PSAP. All the MF spill information goes in the SIP From: header in one string.

The PSAP operator may transfer the call by sending a hookflash (a SIP INFO message with a "hookflash" body). The gateway converts that to a wink toward the tandem, preparing it to receive DTMF digits for the phone where the call should be transferred, sent in a separate message. A SIP REFER message from the PSAP, addressed to the gateway's IP in the Refer-To URI, will transfer the call to another extension at the PSAP whose number is in the "userpart" of the Refer-To URI.

When the VoIP call in the PSAP ends, the IP phone sends a BYE to the gateway. It then reverses the polarity toward the tandem switch, returning to idle.

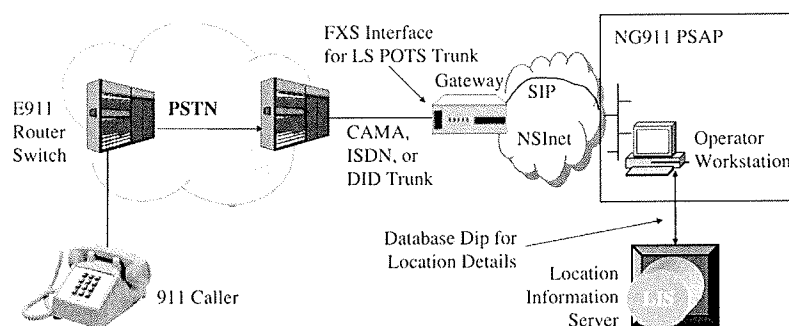


FIGURE 6.1 Gateway to receive a 911 call from the PSTN switch presents an FXS interface to appear as another PSTN switch.

6.3 FACSIMILE TRANSMISSION

Faxing remains important as a way to deliver legally recognized signed documents. Faxes are difficult to alter. By law they should be marked with the sender's name, phone number, and date stamp. A UC application can deliver faxes to individual email boxes and track in- and out-bound faxes for auditing purposes.

Unfortunately, every fax over IP implementation seems to differ in some way from every other. Hosted PBX and SIP trunking providers may handle fax calls the same as voice, using PCM encoding of the modem tones. Or they may decode the modem into a digital stream, carried in packets. Either way, they usually charge about \$10/month for a fax DID number, which may be a dedicated MGW or the second port on a voice MGW.

Fortunately, the SIP Forum spent most of 2010 analyzing facsimile connections and came up with recommendations for changes to T.38. ITU is working on a draft of a revised T.38 at the same time. With the problems more clearly understood, it is highly likely that SIP interoperability testing will sort out a solution in the near future.

6.3.1 Facsimile on the PSTN

ITU's Recommendation T.30 describes real-time faxing over the public switched telephone network (PSTN) between two fax machines with analog (modem) interfaces. It's a universal format, the primary standard for almost all fax machines. The procedure is a dial-up connection originated by one machine and answered by the other. They recognize each other as fax machines by the tones they play toward each other. T-30 is digital at its core but most often travels as analog modem tones. A modem converts the digital fax format to an audible sound that passes like voice through a dial-up connection on the PSTN. Modems imitate voice in a way that PCM can reproduce.

Recall that the PSTN has practically no jitter and creates no interruptions to the audio path. Modem designs assume those conditions, which are not true for VoIP or IP traffic. Fax images, in particular, assume a dedicated voice channel for its format of a constant stream of pixel data.

T.30 persisted when communications technology converted to digital because modems were designed when channel banks had already digitized parts of the PSTN—fax machines were designed for the digital network. Almost always on the PSTN the voice channel remains uncompressed, using PCM/G.711 end to end, in part to ensure success of fax transmissions.

Carriers that compress voice can ensure success in faxing with fax detectors that recognize the modem tones and prevent compression. In a SIP environment the media gateway that connects a fax machine to an IP network should recognize the fax modem tones and re-INV to change the session from voice to T38fax. Problems exist because T.38 isn't precise on which end should invoke the change.

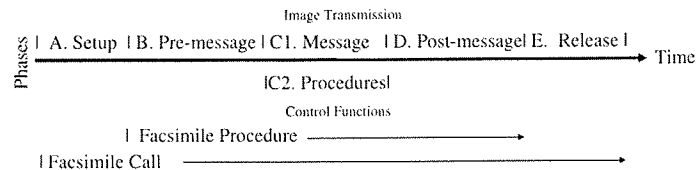


FIGURE 6.2 Phases of a fax transmission.

Fax-aware gateways respond to the difficulties with modems over IP by extracting the scan data (demodulating the modem tones) and transmitting digital information. The T.30 process assumes that the initial scanning to a digital signal follows ITU Recommendation T.4, which also includes run-length compression of blocks containing consecutive white or black pixels on a scan line.

On the WAN connection, T.30 puts the control information in HDLC frames marked by the 0x7E flag character (01111110) that is not used elsewhere. The T.4 data stream from a scan is not framed but runs continuously. Each scan line ends with a special EOL symbol (000000000001) that cannot appear anywhere else in a message. Between EOLs is a series of run-length codes, alternating between black and white, indicating a number of contiguous pixels of the same color. Here lies the sensitivity to jitter and lost packets. If the transition from black to white is confused, the received page can be illegible. On a voice channel the transmission “resets” after each line of pixels. When packetized, catching an error such as a lost EOL symbol is harder.

Transmission via modem takes place in phases (Figure 6.2) of a T.30 connection:

1. Call setup/clearing, such as dialing across the PSTN.
2. Facsimile procedures or commands by which the fax machines recognize each other and select formats.
3. Message transmission, an image encoded per T.4 or other data, sent continuously.
4. Confirmation of receipt.
5. Call termination.

Moreover fast modems have internal echo canceling, which can't train if there is even a small amount of jitter. A dynamic receive jitter buffer still allows enough jitter to pass through to spoil the connection. Network managers should turn off the dynamic adjustment feature for jitter buffers handling fax channels. Turn off VAD as well; some VAD implementations may ignore a modem tone and emit silence instead. That means end of page to the receiving fax machine, and end of transmission.

The slower modems don't cancel echo themselves, so they may be able to communicate as VoIP without resorting to T.38 gateways. However, the time to send a page increases significantly.

Greater problems for fax emerge when the voice carrier transcodes the originating PCM format to compress it into something that requires less bandwidth. ADPCM at 32 or 16 kbit/s has been used for international connections for decades. Two conversions in the path (slower, then faster) distort the reproduction of full-speed modem tones and usually make them unintelligible to the receiving fax machine. Most machines can shift to simpler modulation schemes, at slower bit rates, which may allow a transmission to succeed, but slowly.

The delay in a satellite hop is another problem, which some machines can overcome. Adding the delay of a packet network to a satellite hop make the case more difficult. Latency over 3 s (the so called T4 timer) may trigger retransmissions that overlap with delayed packets. Good spoofing of the T.30 protocol allows a fax relay system to operate with 5 s total latency.

If this is confusing, don't feel bad. Fax machines use multiple modem modulation schemes, from V.21 at 300 bit/s to V.17 to V.34bis at up to 33.6 kbit/s—on each call. The procedures and tone frequencies changed when the Recommendation was updated in 1996. Machines built to that earlier standard should be out of service by now.

Faxing over a packet network (via VoIP) offers three major scenarios under current standards:

- Real-time transmission or fax relay (T.38).
- Store and forward fax transmission (T.37).
- Revert to T.30 for the WAN portion of a connection.

6.3.2 Real-Time Fax over IP: Fax Relay or T.38

ITU Recommendation T.38 addresses the problem of how to fax in real time over a packet network by defining gateways and a protocol between them. This network replaces the audio tones in a voice channel with the digital run-length codes of the scanned pages.

Real-time transmission most closely emulates the analog operation over the PSTN. The users know they have concluded their transaction. The communications carrier never “holds” the information, so never has liability for its security under laws about health information (e.g., HIPPA) or privacy of personally identifiable information.

On the hardware, T.38 describes an interface to the packet network that carries the digital information from the scan output, defined in Recommendation T.4, in packets. A very few fax machines have a digital T.38 interface.

Almost all ordinary fax machines connect in analog (T.30 mode) on a modem interface designed for the PSTN. Rather than send that audible signal over a PSTN connection via modem, a T.38 process takes the modem tones into a device (a gateway) that demodulates the T.30 modem traffic into digital image information. The T.30 data stream then breaks into data blocks that are

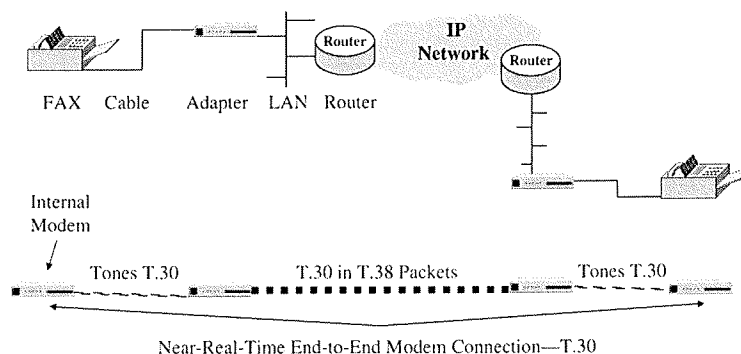


FIGURE 6.3 Real-time transmission of facsimile over and IP network (T.38).

encapsulated in a special version of UDP (with error correction), then in IP packets. A digital network such as a LAN or WAN (Figure 6.3) delivers the fax stream to another fax gateway.

A modulation function in the gateway at the delivery end recreates the modem tones from the transmitted data for the receiving fax machine.

The PSTN offers minimal latency and introduces practically no jitter on an audio channel. To isolate the fax machines from the variable delays and possible packet loss of a packet network like the Internet, a fax gateway will spoof at least part of the T.30 protocol. That is, the gateway terminates or generates the calling and answering tones of the fax call setup procedure (Phase A in Figure 6.2) and accepts DTMF dial strings, emits ringing voltage, and so forth. The packetized page scan content may be protected by optional end-to-end error correction.

Lacking T.38 gateways, fax over VoIP may be treated as voice at “normal” speed (pages per minute) if using the G.711 (PCM) codec. This encoding generates 64 kbit/s plus packet headers to carry a 9.6 kbit/s modem signal—ugly, and still not guaranteed reliable. Even an all-PCM fax connection (no compressions) may fail if any part of the connection is packetized, as in VoIP, without good QoS. Nominally able to carry the modem signal, G.711 over IP often suffers enough from packet loss and jitter to make fax transmission uncertain. This is a concern because the majority of international calls in 2010 were carried as VoIP.

Codecs that compress voice channels cannot carry high-speed modem tones accurately, so the fax machines back down to simpler (and slower) modulation schemes. They too may fail, for the same reasons.

Because the fax machines see only the analog modem signals, a T.38 gateway to the LAN needs to operate in real time, or very near it. That is, the gateway creates packets at short intervals when connected to a sending machine and reproduces a constant analog signal when delivering to the receiving fax. Silence causes the receiving fax machine to hang up.

When T.38 first appeared, RTP wasn't established. A form of UDP called UDPTL (UDP Transport Layer) was standardized and widely implemented—at this time it is still the dominant protocol for T.38 implementations. Version 3 of T.38 emphasizes RTP as the better protocol.

Each UDPTL protocol data unit (PDU) contains a sequence number followed by the current or primary data block—an Internet Facsimile Protocol (IFP) message—from the fax scanner. An IFP message contains an internal sequence number. The two numbers must be the same.

If error correction is configured, there are two options:

- In redundant mode, copies of one or more earlier IFP messages follow the current IFP message in the packet. With the current and the previous interval in each packet, two consecutive IP packets must be lost to lose any scan data. With two earlier IFP messages in each packet, a data loss requires three missing packets, and so forth.
- In forward error correction (FEC), each UDPTL packet contains the primary IFP message followed by a field that indicates the number of earlier messages that are included in the FEC process, then a parity-encoded representation of those messages.

IFP messages processed for FEC must have contiguous sequence numbers, starting with the sequence number just before the primary IFP message. That is, if the primary IFP message is number S , the sequence number of the first redundant message is $S - 1$ of the second, $S - 2$ and so forth.

The clever FEC process creates one composite message equal in length to the longest message included in the group. The previous messages are stacked vertically, the shorter ones are padded with 0's to the length of the longest, then each bit-wide column is "parity checked": if the number of "1" bits in a column is odd, the new message has a "1" in that position; otherwise, the result is "0" for that bit position.

With this parity information from N packets of FEC fields (sent in N packets), the receiver can recreate one missing packet (one primary message) in N . There is also a more complex FEC process that generates multiple FEC messages per IP packet. This scheme protects against some error bursts that lose multiple consecutive IP packets but requires more processing power.

FEC is optional. A gateway receiving redundant IFP messages may ignore them.

To identify a fax connection, the media type in an SDP block may read:

- $m = \text{audio/t38}$, indicating the transmission emulates fax-modem connectivity.
- $m = \text{image/t38}$ when the data is a TIFF file containing an image of the faxed page.

Because of the wide use of UDP and UDPTL, a vendor contribution to the IETF draft of proposed revisions to T.38 offers an example of what a fax-only INV might look like between T.38 gateways (Figure 6.4).

```

INVITE sip:+1-212-555-1234@bell-tel.com SIP/2.0
Via: SIP/2.0/UDP kton.bell-tel.com
From: A. Bell <sip:+1-519-555-1234@bell-tel.com>
To: T. Watson <sip:+1-212-555-1234@bell-tel.com>
Call-ID: 3298420296@kton.bell-tel.com
CSeq: 1 INVITE
Subject: Mr. Watson, here is a fax
Content-Type: application/sdp
Content-Length: ...
v=0
o=faxgw1 2890844526 2890842807 IN IP4 128.59.19.68
e=+1-212-555-1234@bell-tel.com
t=2873397496 0
c=IN IP4 128.59.19.68
m=image 49170 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
m=image 49172 tcp t38
a=T38FaxRateManagement:localTCF

```

FIGURE 6.4 INVITE message for fax connection.

The first “m=” line indicates a preference for UDPTL over TCP, which appears in the second “m=” line. The response will indicate port=0 for the rejected option and a valid port number for the accepted transport. The UA software takes care of creating the message, including the Content-Length field, which was not done for this hypothetical message.

For an INV asking for RTP/UDP, the message could resemble that shown in Figure 6.5. The first “m=” line now contains RTP with the Audio Visual Profile.

After establishing an audio connection between two UACs, one of them will re-INVITE with T.38 as an attribute, moving the call to fax mode based on PCM encoding (G.711) and no silence suppression (no VAD). The SDP message in the body of the re-INV could look like Figure 6.6.

The language of Recommendation T.38 leaves some ambiguity, which allowed different implementations of fax gateways. In some circumstances they don't interoperate. For example, T.38 isn't perfectly clear on which end of a sip connection should issue a re-INV to change from voice mode to fax mode. While gateways can detect fax modem signals of various kinds, only the HDLC flag characters in the initial V.21 modem modulation will distinguish a fax machine from a data modem.

Some T.38 implementations default to a behavior where the sending MGW waits for a media packet from the receiver before starting to send page images. It's like a wink-start telephone trunk, which can be good but raises a problem when working through a firewall with NAT. The media port won't be open to that first packet from the receiver until the sender opens it with a packet from the inside. To work in this case, the sender should be configured to start sending

```

INVITE sip:+1-212-555-1234@bell-tel.com SIP/2.0
Via: SIP/2.0/UDP kton.bell-tel.com
From: A. Bell <sip:+1-519-555-1234@bell-tel.com>
To: T. Watson <sip:+1-212-555-1234@bell-tel.com>
Call-ID: 3298420296@kton.bell-tel.com
CSeq: 1 INVITE
Subject: Mr. Watson, here is a fax
Content-Type: application/sdp
Content-Length: ...
v=0
o=faxgw1 2890844526 2890842807 IN IP4 128.59.19.68
e=+1-212-555-1234@bell-tel.com
t=2873397496 0
c=IN IP4 128.59.19.68
m=image 49170 RTP/AVP 100 101
a=rtpmap:100 t38/8000
a=rtpmap:101 parityfec/8000
a=fmtp:101 49173 IN IP4 128.59.19.68
a=T38FaxRateManagement:transferredTCF
m=image 49172 tcp t38
a=T38FaxRateManagement:localTCF

```

FIGURE 6.5 Connection request for RTP/UDP in a fax connection.

```

v=0
o=faxgw1 2890844526 2890842807 IN IP4 128.59.19.68
s=FAX message
e=faxsupport@company.com
t=2873397496 0
c=IN IP4 128.59.19.68
m=application 49170 udp t38
a=t38errctl:parFEC

```

FIGURE 6.6 SDP message in the body of a re-INVITE.

blank or idle packets (also call “no-op”) immediately on the port of the image session. That will open the port on the firewall for packets from the sender.

The SIP Forum has a working group on the task of improving fax over IP. A key goal is to define repeatable tests to verify interoperability and capacity to handle multiple FoIP sessions, which usually require more CPU cycles than a voice session.

Version 3 of T.38 was nearing completion in 2011. It increases the maximum speed of transmission (lowers the time per page) by standardizing on a V.34 modem, described in the ITU Recommendation of that number, which is faster (up to 33.6 kbit/s). Older machines are built on V.17 modems for page images, 14.4 kbit/s maximum. When selecting a fax machine, fax server, or media gateway with fax capability, look for the T.38 Ver. 3 capability but be sure it can fall back gracefully to V.17 modulation.

6.3.3 Store-and-Forward Fax Handling

To automate handling of faxes, the receiving process can be emulated in software on a server equipped with a fax modem. Calls from the PSTN arrive over analog or ISDN lines. The modem demodulates received fax tones into a digital form that the server converts to a file format (typically TIFF) for delivery, storage, and viewing.

If the fax server is owned by the recipient, the server is considered the end point and the arrangement is nearly real time. If a fax server is in the middle of the network—outsourced to the carrier or a third party—the message transfer may become a “store-and-forward” process described by T.37 (Figure 6.7). It is more like email for faxes. The sending fax machine transmits to a server, in either T.30 or T.38 format. The server puts the digital information on disk, where it may sit until the recipient asks for it. Or the server may deliver the message to an outbound gateway that dials a destination fax machine over the PSTN and sends the fax via the T.30 procedures.

A server holding faxes may be subject to compliance and security requirements. Like emails, stored faxes may be archived for auditing purposes and to satisfy requirements for legal discovery. The server can offer additional security by authenticating users who try to retrieve faxes, encrypting fax files, and logging when and by whom faxes are originated and retrieved.

The server may deliver the fax at a scheduled time, immediately after receiving it, or on demand. With the fax in a stored file, it is available as part of a file system or the server can forward it as an attachment to an email.

When fax servers email the image files to the user, there’s a MIME type for that (AUDIO/T.38) so that the form of the attachment is understood. The recipient can then pick up the fax like any email. Other delivery options include converting the fax text to spoken words when the recipient calls in, and delivery to a specified printer.

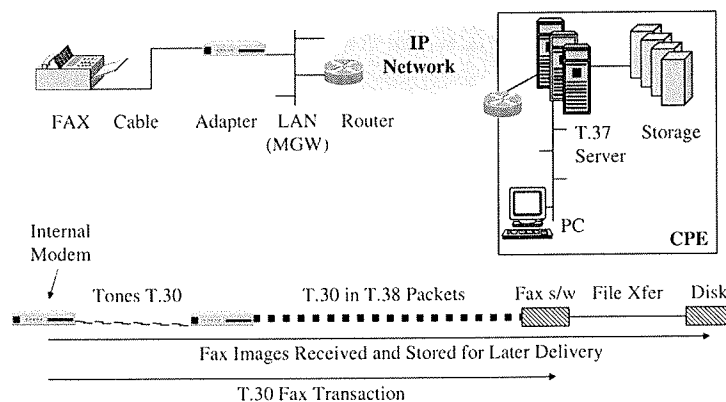


FIGURE 6.7 Fax via a store-and-forward server on a packet network (T.37).

Digital local loops such as ISDN lines will identify the called number with each inbound call. Caller ID and DID services are also offered on analog POTS lines. The fax server can associate a fax with the intended recipient through those numbers for delivery to an email address or a voice mail box.

Hosted fax services boast of large capacity to receive simultaneous faxes to the same phone number. An enterprise need not dedicate a large number of lines or machines to receiving faxes in response to a temporary rush such as a closing date, special product sale.

6.3.4 IP Faxing over the PSTN

So far, fax over IP as part of Unified Communications sounds fairly straight forward. However, the standards are not always interpreted identically by all vendors of equipment and software. There have been incompatibilities, though they are diminishing. Even the most popular brands of Fax to IP adapters don't work reliably enough for some critical applications.

The most common problems arise from the sensitivity of fax machines to delay and lost information. Too much of either and a fax machine can hang up and issue an error report.

When connected over a dialup connection on the PSTN, a voice channel has lower latency and practically no jitter compared to a packet network. IP transmission almost always increases latency (from buffering packets at multiple routers and switches across the network). If the packet network is congested, latency can vary (variation in latency is jitter).

Fax/IP systems that face regular fax machines need to look like another regular fax machine connected on the PSTN, a form of "spoofing." That is, a fax gateway needs to respond quickly and consistently to the signals from a fax machine, while dealing with a packet network that can introduce jitter and information loss.

As of this writing, the bullet-proof fax system is based on keeping the Fax over IP transmission on a LAN where the quality of service is controlled to a high level. Off-premises faxing reverts to the PSTN.

Hardware consists of fax modem cards in a server on each LAN. The POTS interfaces on the modem cards connect to local trunks. Workstations at both sites may run fax software, obtaining the benefits of message logging, easily viewable files, and flexible delivery. But each site uses the gateway to the PSTN for off-site faxing (Figure 6.8). The sending side dials up a voice connection and sends modem tones (T.30). The receiver may be another fax server or a standard fax machine.

This architecture certainly isn't elegant, or all-digital, and it doesn't rely on the Internet for inter-site transport. Many designers will reject it as inappropriate to include in a "UC" solution. However, the facsimile protocol in this case operates only over the PSTN, as it was designed. Users get all the UC benefits of routing and delivery to and sending from the desktop.

The optimal choice for you depends on how the business and the people use fax transmissions.

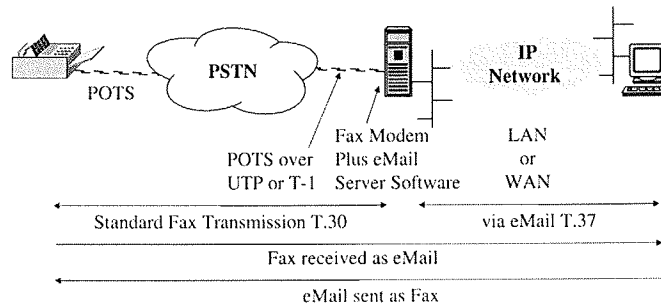


FIGURE 6.8 Sure-fire fax handling connects directly to the PSTN for off-site communications.

6.4 PHONE FEATURES ADDED WITH VoIP/UC

More than 50 projects in the IETF, additional ones in the SIP Forum, and proprietary efforts reminiscent of the PBX feature wars keep adding to the functions available on new telephone systems. Unified Communications integrates many additional features such as video, presence, and instant messaging. There are more than enough to distract the potential buyer.

The focus, then, must be on what's essential to your way of doing business. Review of sales literature will say more about emerging features. For starters, here are some things that should be available from most sources.

- **Coupling the PC and the phone:** manage the phone and calls on a PC. See who's calling, place calls, record calls, set up conferences from a web browser window or calling application.
- **Complete call lists:** not just completed calls, but missed or abandoned inbound calls—return the call with a mouse click.
- **Company and personal directories:** extensive information about fellow employees, provided from the HR database. Your contacts entered with space for as much information as you want from virtual business cards.
- **Privacy from integrated encryption:** IP phone participates in a virtual private network (VPN), IPsec, or Secure RTP—perhaps all three plus other methods—and can be configured and invoked easily from the phone or browser.
- **Choice of codecs:** for voice and video. Provides a range of trade-offs for bit rate versus voice quality.
- **Facsimile:** with a public telephone number (E.164), receive faxes in your private email box. Retrieve them from any browser or an application on a mobile phone or desktop computer. Send documents and images to any fax machine. For more details, see Section 6.3, Facsimile Transmission.

6.4.1 Presence

A survey (by Rad) revealed that end users consider presence very important in the business environment. This is not just for instant messaging (are you logged in?), but based on any information available about a user. At this writing, systems can display the busy status of the phone and meetings scheduled in calendaring servers. One vendor claimed to update presence if the keyboard is used at all.

In VoIP, status comes from the server where a user agent registers. This could be a hosted service or an “owned-and-operated” server. Many companies operate their own IM servers, for confidentiality, message archiving, or other reasons. The IM server can also track presence.

6.4.2 Forking

Find me/Follow me service in legacy switches would ring a series of telephone numbers until one answered or the call reached voice mail. Because numbers rang several times at each phone before the “ring no answer” count tripped the move to the next number, it could take significant time to run through the list.

VoIP systems improve on this feature by “forking” a call to ring multiple phones at the same time. The first to pick up gets the call. The phones that don’t answer quickly return to idle status so that they can send or receive other calls—they are not tied up until the answered phone goes back to the on-hook state.

Forking proved handy in early VoIP deployments and became a favorite of people who weren’t hiding behind voicemail. The caller got through whether at an office desk phone, cell phone, home phone, car phone (remember them?), or the girl friend’s phone.

However, a proxy that forks an INVITE to multiple UASs will not do so unless the calling UAC provides acceptable authentication credentials for all of the UASs that ask for them. Smart proxy servers can still apply filters of various kinds to block or send to voicemail calls from specific sources, at defined times, or on designated days of the week.

6.4.3 Voicemail = eMail

Some organizations live in voicemail, some in email. UC promises to merge the two.

The technology to convert voice mail to text is speech recognition. It is highly developed in over 40 languages. When a caller leaves a voicemail, a UC system can convert it to text and deliver it via email. With cell phone displays now big enough to read the text, this can be a good way to retrieve messages in noisy areas such as airports or sports arenas.

Conversely, an email or text message converts to a voicemail via the text-to-speech feature.

The original message remains available in its first format so the recipient can check words that didn’t translate or seem suspect from the context. Vendor

offerings vary, so check for convenience, reliability of conversions, and costs in terms of delivery latency and required servers.

6.4.4 SMS Integration

Short Message Service (SMS) was part of the GSM cell phone technology from the start. SMS is better known as “texting.” Teens live by SMS, but big business has found applications such as alerting customer to delayed air flights, low bank balance, or items on sale. An interesting possibility suggested by an SMS vendor is to update an employee’s presence status, including call forwarding, through a text message to a directory server.

The background on SMS involves Signaling System 7, which has a “part” for mobile users to send text messages. The destination typically is another mobile phone. Like most telco protocols, SMS was designed to be compact—the payload is limited to 140 octets on the GSM control plane. If using 7-bit ASCII codes, the maximum number of characters rises to 180. Complex character sets (Chinese, Japanese and Korean, Cyrillic) require 16-bit Unicode so only 70 characters fit in the SMS packet. Yes, that includes the title and space characters.

Extensions and services added by carriers and entrepreneurs has grown into a huge volume of traffic. In 2010 the number of text messages exceeded the number of phone calls on cellular system.

Twitter is based on SMS messaging. Carriers offer delivery of email via SMS, where up to the first 1600 octets of the email message is split over multiple SMS packets. Sprint will deliver an SMS message to a PSTN phone via automated text-to-voice conversion. If not on a plan with unlimited text messages, the charge of \$.15 each can add up quickly.

The specialized nature of SMS has been a barrier to integrating the service in business processes. As mentioned, carriers will convert and deliver a text message in other forms. Specialized service bureaus or data brokers also offer ways to connect computers (e.g., of call center agents) to SMS. Unified Communications controllers can treat SMS as just another medium. Incoming texts can be queued at a call center with email and voice calls, to answered in order.

A common interface to send and receive SMS texting is as an email. The conversion requires a gateway server, these days connected on an IP network. Formerly a modem link was part of the infrastructure, but it’s no longer is needed when HTML is the transport.

Only a few of the more sophisticated call control software platforms integrate SMS. At this time applications called Twilio and Voxeo Prophecy have the capability to send and receive SMS. FreeSwitch can receive but not send SMS. Asterisk, one of the VoIP software packages that’s been around the longest, doesn’t handle SMS at this time.

Because SMS is essentially a service of cellular carriers, it is harder to set up private services as is possible with instant messaging.

6.4.5 Instant Messaging

IM educated users about the concept of presence. The IM application on your PC displays your contacts who are on line at the time, as you are shown to them. Almost 6% of respondents to a survey in 2011 indicated this function is important to their businesses.

To distinguish IM from SMS, the difference is in the transmission method and the kind of server that mediates the service. An enterprise may host the IM server if it wants to control its internal communications.

- IM relies on a server where users register to obtain an account, then log in to establish their presence. The server may reside at AOL, Yahoo, another third-party data center, or on premises.
- SMS is a carrier function, since it relies on the signaling system of the cellular networks. SMS “brokers” may transfer messages between the cellular side and email or another form of message transfer. This function is integrated, for example, into comprehensive call center software.

IM can attach files, and messages can be longer than on SMS. SMS doesn’t necessarily report on presence for your contacts.

The protocol of IM, XMPP, *Extensible Messaging and Presence Protocol* (RFC 6120), originated as the Jabber protocol about 1999. XMPP uses streams of XML (eXtensible Markup Language) to convey information, request responses, and maintain presence. Streams may be encrypted as TLS or authenticated (using a Simple Authentication and Security Layer, SASL) mechanism within this protocol.

IM servers authenticate themselves to each other, but each server has the responsibility to authenticate its own registered users. An IM server therefore cannot vouch for the name or address of a user registered elsewhere. Address imitation and forgery are possible.

XMPP addresses can be in any properly coded character set. Beyond the simple faking possible by substituting a numeral one (1) for a lowercase letter L (l), a user can switch character sets to be more creative with “confusable characters” in imitating a trusted address or name.

Unfortunately, there seems to be no technical way to prevent deliberate confusion. An IM server can make fakes harder by limiting all names to one font and character set (see Figure 6.9).

IM relies on XML over TCP. Signing on to an IM server starts preferably with a TCP connection negotiated with Transport Layer Security (TLS) for channel encryption; other transport is possible.

On that connection the client and server open an XML stream. In a sense the procedure is the same as starting to transfer an XML document. The two ends bind their specific resources to an XML stream. However, rather than closing the connection after a file transfer, the XML “document” remains “partly transmitted” for as long as the user is logged in.

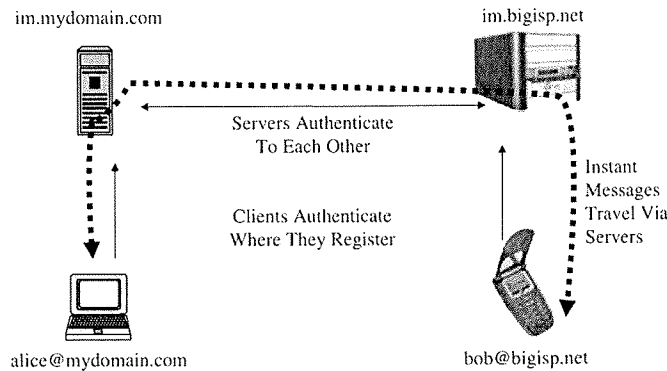


FIGURE 6.9 Instant messaging servers authenticate locally and between servers when accepting and delivering messages.

Adding an XML session fits easily with TCP as both are stream oriented. Each end of the XML connection binds a resource, like an application, to the stream. TCP and XML connections remain open indefinitely, allowing either party to push information to the other at any time.

When connected to the IP address and port of the receiver, the initiator opens a stream by sending the “initial stream header” to the receiving entity:

```
<?xml version='1.0'?>
<stream:stream
  from='juliet@im.example.com'
  to='im.example.com'
  version='1.0'
  xml:lang='en'
  xmlns='jabber:client'
  xmlns:stream='http://etherx.jabber.org/streams'>
```

The receiver replies with the “response stream header”:

```
<?xml version='1.0'?>
<stream:stream
  from='im.example.com'
  id='++TR84Sm6A3hnt3Q065SnAbbk3Y='
  to='juliet@im.example.com'
  version='1.0'
  xml:lang='en'
  xmlns='jabber:client'
  xmlns:stream='http://etherx.jabber.org/streams'>
```

The entities then finish the stream negotiation process, which must include authentication via credentials using SASL. Servers may require encryption

(TLS). The order is TCP, TLS, SASL, then XMPP. For enhanced security, the devices can “forget” some of the information learned during the negotiation: the client and server may flush memory by exchanging new stream headers.

Headers may include “features/” elements, listing what is required or must be negotiated. Certain features require a restart when negotiated, for example, TLS, which is done by sending new stream headers on the same TCP connection. The XML stream gets a new ID number but is not closed.

Communications takes the form of blocks of information called XML stanzas. Any number of stanzas may be exchanged over the XML stream. There are three kinds of stanzas:

- **Message:** a push capability to send data,


```
<message to='foo'>
  <body/>
</message>
```
- **Presence:** publish-subscribe mechanism to advertise availability on the network,


```
<presence>
  <show>
</presence>
```
- **iq (Info/Query):** request–response mechanism similar to HTTP.
 - Request:


```
<iq to='bar'
  type='get'>
  <query/>
</iq>
```
 - Response:


```
<iq from='bar'
  type='result'>
  <query/>
</iq>
```

These three elements are allowed only after completion of the stream negotiation. If either end attempts to send on earlier, the receiver must reject it and close the stream.

Users can send and receive any number of XML stanzas with any other users on the network. When a user signs off, the IM client closes the XML stream first, by sending a “stream/” tag, and then closes the TCP connection after confirming the XML closure. This procedure prevents opening a security vulnerability which arises if the TCP connection closes first.

The terminals on a long-lived TCP connections may not notice that it has failed during a period of inactivity. Therefore a stanza may be lost.

The “xmpp-client” uses port of 5222 for client-to-server connections. Port 5269 is standardized for server-to-server connections. These are the default ports registered with IANA).

6.4.6 Webinar Broadcasts

Lots of webinars in late 2010 replaced the media/analyst/consultant tours that vendors previously had to make when announcing a new product or company. The time and travel budgets saved certainly helped in the Great Recession. In many ways the web meeting is better for the recipient of information, too. For one, the recorded meeting can be replayed later to confirm details. Vendors use replays for internal training and promotional pieces.

With just two parties on a web conference, they can collaborate on writing or editing a document, create slides, or examine and discuss a complex chart or diagram in detail. It's more than a video conference.

6.4.7 Telepresence

Think video conferencing, but with feeling. The concept is to duplicate a meeting "across the table" by applying high technology and psychology. The images are hi-definition, the audio is hi-fi (7,000 Hz at least), and the lighting, camera angles, and seating are designed to contribute to the feeling of being there. Stereo sound places the speaker's voice near his image.

Broadband connectivity and H.264 video compression make the technology practical; only a half Mbit/s is needed. Dramatic drops in pricing for the components make it practicable.

Standardization proved a boon to telepresence. Different vendors' equipment interoperates. There is no need to keep the participant list to those inside one organization.

Global networks and global business leverage the savings from time and travel expense. One report claims a complete payback on a video conferencing investment in seven days of training sessions for people across the world.

A major push for the use of telepresence in the medical field shows promise of making doctors, particularly specialists, available to rural and remote areas. Thinly populated regions can't support all the specialties all the time. Doctor on demand quickly responds to a need without incurring the expense of full-time staffing at all locations.

6.4.8 More UC Features to Consider

Each UC vendor wants a unique selling proposition to convince prospects to buy its version. Look for the differentiators among features that can help your business, which could include:

- Real-time collaboration at a distance, between two users or across a group on a conference call (multicast connections). Meetings may be ad hoc, or configured administratively for a fixed team. Avaya features a drag-and-drop ability to pick participants from a directory listing to add people to a meeting quickly.

- In addition to audio conferencing on demand, UC that offers white board sharing, co-editing documents, photo distribution, file sharing, and prepared video footage as well as the speaker's image.
- Mobility whereby several vendors offer complete VoIP/UC phone features on portable devices such as netbooks and tablets as well as laptops and smartphones.
- Automated directories that combine all employees with contractors, vendors, and other contacts such as press and analysts.
- Instant Messaging integrated with system services, allowing side messaging during a conference call to any or all participants.
- Calendaring and scheduling playing an important role as at some firms where every meeting request comes through an invitation for a specific slot on the calendar. This can be too formal for some users, but it does make it clear when people are not available.
- Dashboard display of system status including usage, presence, and the modern busy lamp field.

DDM-4



FCC Requirements for Interconnected VoIP Providers

Prepared by John Staurulakis, Inc. – Updated August 2016

I. Definition of VoIP Providers Subject to These Requirements (Section 9.3 of the FCC's Rules)

These requirements apply to “interconnected VoIP providers” which means that they meet the following criteria:

- enables real-time, two-way voice communications;
- requires a broadband connection from the user's location;
- requires IP-compatible customer premises equipment; and
- permits users to receive calls from and terminate calls to the PSTN

II. 911 Requirements (Section 9.5 of the FCC's Rules)¹

A. Subscriber Notification, Acknowledgment and Labeling Requirements (went into effect on July 29, 2005) - Any new provider of interconnected VoIP service must:

- Specifically advise every new subscriber, prominently and in plain language, of the circumstances under which E911 service may not be available through the interconnected VoIP service or may be in some way limited by comparison to traditional E911 service;
- Obtain and keep a record of affirmative acknowledgement by every subscriber of having received and understood the advisory described in the paragraph above; and
- Distribute to each new subscriber warning stickers or other appropriate labels warning subscribers if E911 service may be limited or not available and instructing the subscriber to place them on or near the equipment used in conjunction with the interconnected VoIP service. The warning stickers or other appropriate labels should be distributed to each new subscriber prior to the initiation of that subscriber's service.

(If you would like assistance in drafting language to use in your service agreements, notices, acknowledgment forms or labels, please let us know).

¹ Note that the FCC has requirements for VoIP providers that exceed a certain revenue threshold to submit 911 and E911 analyses and reports pursuant to Section 12.3 of its rules. The revenue requirements are the same as those which distinguish mid-size and rural telephone companies (“Class B Companies”) from larger ILECs (“Class A Companies”) found in Part 32.11 of the FCC's Rules. Accordingly, small VoIP providers likely would not be required to submit the reports. Please let us know if you would like for us to make the calculations to determine if your company exceeds the thresholds.



B. Provide 911 Service (went into effect on Nov. 28, 2005). Each interconnected VoIP provider must:

- Transmit all 911 calls to the public safety answering point (PSAP), designated statewide default answering point, or appropriate local emergency authority that serves the caller's "Registered Location."² Such transmissions must include the caller's Automatic Numbering Information (ANI)³ and Registered Location to the extent that the PSAP, designated statewide default answering point, or appropriate local emergency authority is capable of receiving and processing such information;
- Route all 911 calls through the use of ANI and, if necessary, pseudo-ANI,⁴ via the Wireline E911 Network,⁵ and make a caller's Registered Location available to the appropriate PSAP, designated statewide default answering point or appropriate local emergency authority from or through the appropriate Automatic Location Identification (ALI) database;
- Obtain from each of its existing and new customers, prior to the initiation of service, a Registered Location; and
- Provide all of their end users one or more methods of updating their Registered Location at will and in a timely manner. At least one method must allow end users to use only the same equipment (such as the Internet telephone) that they use to access their interconnected VoIP service.
- In all areas where the carrier is not able to transmit 911 calls to the appropriate PSAP in full compliance with the Commission's rules, the carrier must not market VoIP service or accept new customers for their service.

(Please let us know if you would like our assistance in determining if your company can comply with these requirements or to determine if seeking waiver of one or more of the requirements is necessary).

² An end-user's "Registered Location" is the most recent information obtained by an interconnected VoIP service provider that identifies the physical location of the end-user.

³ ANI is a system that identifies the billing account for a call and, for 911 systems, identifies the calling party and may be used as a call back number.

⁴ Pseudo-ANI is "a number, consisting of the same number of digits as ANI that is not a North American Numbering Plan telephone directory number and may be used in place of an ANI to convey special meaning. The special meaning assigned to the pseudo-ANI is determined by agreements, as necessary, between the system originating the call, intermediate systems handling and routing the call, and the destination system."

⁵ The "Wireline E911 Network" is a "dedicated wireline network that: (1) is interconnected with but largely separate from the public switched telephone network; (2) includes a selective router; and (3) is utilized to route emergency calls and related information to PSAPs, designated statewide default answering points, appropriate local emergency authorities or other emergency answering points."



III. CALEA Requirements (Section 1.20000 and following of the FCC's Rules)

A. Compliance Deadline

- As of May 14, 2007, all facilities-based broadband Internet access providers and interconnected VoIP providers must ensure that their equipment, facilities or services are CALEA compliant.
- Interconnected VoIP providers that are resellers of VoIP services should make sure that their underlying VoIP provider is CALEA compliant.
- Facilities-based interconnected VoIP providers should talk to their vendors and other manufacturers to determine what needs to be done from a technical standpoint to comply.
 - The FCC recognizes that some interconnected VoIP providers such as those who plan a nationwide deployment will need to incorporate a CALEA solution into numerous routers or servers or negotiate arrangements with numerous interconnecting carriers.
 - FCC allows interconnected VoIP providers the alternative to purchasing and installing all necessary equipment and performing all functions in-house is using "Trusted Third Parties" (TTPs). Under this approach, the TTPs operate as a service bureau with a system that has access to a carrier's network equipment and remotely manages the intercept process for the carrier. FCC found this to be a "reasonable means" for smaller carriers to comply with CALEA.

B. Systems Security and Integrity (SSI) Policies and Procedures

- Interconnected VoIP providers must develop and file their CALEA SSI Policies and Procedures with the FCC (LECs and other telecom carriers had to file in 2000).
- If an interconnected VoIP provider is affiliated with telecom carrier, either the telecom carrier's SSI policies and procedures can be amended to reflect the VoIP company or the VoIP company can prepare and file a separate document.

(Please let us know if you would like our assistance in preparing and filing new or amended SSI Policies and Procedures).

IV. CPNI (Sections 64.2000 – 64.2200 of the FCC's Rules)

- The FCC ruled that all CPNI requirements for telecom carriers apply to interconnected VoIP providers.



V. FCC Form 499-A Requirements

A. Registration

- Section 1.47 of the FCC's Rules requires interconnected VoIP providers to register with the FCC within 30 days of commencing service by completing pages 1, 2, 3 and 8 of the FCC Form 499-A which includes listing a DC agent for the FCC to serve notices.

B. Contribution to USF

- Interconnected VoIP carriers became subject to USF payments in 2006. Section 54.706 of the FCC Rules requires that annually by April 1st, interconnected VoIP providers report their interstate revenues on the FCC Form 499A. The FCC provides the following alternatives for interconnected VoIP providers:
 - Safe Harbor - the FCC allows interconnected VoIP carriers to determine the interstate portion of their service offerings using a "safe harbor" percentage. Currently, the safe harbor percentage for interconnected VoIP providers is 64.9% of the end user revenue. This percentage is applied to the end user revenue to determine the amount that is interstate and then the relevant quarterly USF contribution factor is applied to this amount.
 - Alternatively, the Interconnected VoIP carrier can report actual interstate revenues (which most VoIP providers are not able to do) or perform a traffic study which then can be used as a proxy to determine the proper percentage of revenue that is in the interstate jurisdiction. Before using a traffic study, however, interconnected VoIP providers must submit its proposed traffic study to the FCC for approval. Traffic studies will need to be submitted quarterly with the 499Q submissions.
- Resellers of interconnected VoIP should coordinate with their underlying VoIP provider regarding the reporting of revenues for the purposes of contributing to USF. Similar to telecom resale, the underlying interconnected VoIP provider is required to list revenues that it receives from resellers on the form and resellers must provide a statement certifying that the carrier contributes to USF.
- VoIP providers can recover the amount that they contribute to USF by a line item charge on the end users' bills; however, the amount charged to the end user must not include a mark-up above the relevant contribution factor.
- VoIP providers whose annual contribution for USF will be less than \$10,000 when calculated using the worksheet on the Form 499-A are considered *de minimis* and are not required to contribute to USF for that year; however, the provider must still complete and submit the Form 499-A.



C. Contribution to TRS

- Section 64.604 of the FCC's Rules requires interconnected VoIP providers to contribute to the interstate TRS Fund by reporting its interstate end-user revenues on Form 499-A which it does when it reports its interstate revenues for the purpose of contributing to USF using one of the methodologies outline above.

D. Contribution to Costs of Establishing LNP

- Section 52.17 of the FCC's rules requires interconnected VoIP providers to contribute to meet the costs of establishing numbering administration by reporting its interstate end-user revenues on Form 499-A which it does when it reports its interstate revenues for the purpose of contributing to USF using one of the methodologies outline above.

VI. Disabilities

A. Adhere to the Same Disabilities Obligations of Telecom Providers (Section 6.1-6.23 of the FCC's Rules)

- These rules include:
 - ensuring that the service is accessible to and usable by individuals with disabilities, "if readily achievable;" and if not "readily achievable," ensuring that the service is compatible with existing peripheral devices or specialized customer premises equipment commonly used by individuals with disabilities to achieve access, "if readily achievable;"
 - ensuring that information and documentation provided in connection with equipment or services be accessible to people with disabilities, where "readily achievable" and that employee training, where provided at all, account for accessibility requirements;
 - maintaining records of the accessibility efforts demonstrating compliance with the disability requirements that can be presented to the FCC in the event that consumers with disabilities file complaints and annually certify compliance; and
 - designating and submitting to the FCC contact information for an agent for service of disability access-related inquiries or complaints.

B. Offer 711 (Section 64.601 of the FCC's Rules)

- The FCC requires interconnected VoIP providers to offer 711 abbreviated dialing for access to relay service to ensure that TRS calls can be made from any telephone, anywhere in the United States, and that such calls will be properly routed to the appropriate relay center.



VII. Numbering (Part 52 of the FCC's Rules)

- Most of the numbering rules apply to interconnected VoIP providers including number portability rules except that an interconnected VoIP carrier cannot directly be assigned numbers.

VIII. Form 477 (Section 1.7001 of the FCC's Rules)

- In June 2008, the FCC revised its Form 477 to require VoIP providers to complete Parts II (B) and V of the form which tracks the number of residence and business lines provided to customers.

IX. Regulatory Fees

- Since 2007, the FCC's Orders establishing the annual regulatory fees have required VoIP providers to pay regulatory fees based on revenues reported on the Form 499-A.

X. Discontinuance or Reduction Filing Requirements (Section 63.60 of the FCC's Rules)

- The FCC requires interconnected VoIP providers to follow the same filing procedures as telecom carriers if the provider intends to discontinue or reduce its service. This process involves notifying customers if the service will be discontinued or reduced and filing an application for discontinuance/reduction with the FCC.

XI. Outage Reporting Requirements

- Effective December 16, 2012, the FCC's Outage Reporting requirements apply to VoIP providers. The FCC released a Report and Order February 21, 2012 which extended mandatory outage reporting rules to facilities-based and non-facilities based interconnected VoIP service providers and applied the current Part 4 definition of "outage" to outages of interconnected VoIP service, covering the complete loss of service and/or connectivity to customers at least 30 minutes duration that potentially affects at least 900,000 user minutes of interconnected VoIP services and results in complete loss of service; or potentially affects any special offices and facilities such as a 911 facility.

XII. Truth in Billing (Not Applicable at This Time)

- Currently, the FCC's Truth-in-Billing rules do not apply to VoIP providers; however, in September 2009, the FCC released a Notice of Inquiry seeking comment on whether these rules should apply to VoIP providers, and on April 27, 2012 released a Further Notice of Proposed Rulemaking requesting comments on "cramming" issues and solutions related to VoIP service. The comment period ended July 9, 2012.



XIII. Call Blocking Prohibitions and Phantom Traffic

- In addressing rural call completion issues, the FCC clarified in the USF reform order released November 18, 2011 and Declaratory Ruling released February 6, 2012 that all carriers are bound by the prohibition on call blocking, including VoIP providers.
- To address phantom traffic, the FCC now requires providers of interconnected VoIP service to include the calling party's telephone number in all call signaling, and requires intermediate carriers to pass this signaling information, unaltered, to the next provider in a call path.

XIV. Access Charges

- In the USF-ICC Order, the FCC clarified that VoIP-PSTN traffic is subject to access charges. An interconnected VoIP carrier must pay terminating access for any toll calls delivered to a circuit switch for termination. However, this terminating VoIP-PSTN traffic is subject to Interstate access rates for both interstate and intrastate traffic. VoIP PSTN traffic originated on the PSTN and terminated to the interconnected VoIP provider is subject to originating intrastate or interstate access charges until June 30, 2014. On July 1, 2014, the originating VoIP-PSTN traffic will be subject to interstate rates.
- To implement access charges for VoIP-PSTN traffic, LECs require the traffic to be identified via a PVU factor. An interconnected VoIP carrier will need to report a PVU factor to the LEC if it is directly connected or to an underlying carrier if it is not directly connected. Most LEC tariffs allow for changes in the PVU factors quarterly.

XV. International Telecommunications Services Reporting

- The FCC's international service reporting requirements apply to VoIP providers. The FCC released a Second Report and Order January 15, 2013 which extended new Section 43.62 rules to entities providing international calling service via VoIP connected to the PSTN. Pure resellers of international calling with annual revenue under \$5 million file a brief streamlined report. Reporting requirements for facilities-based providers are more extensive.

XVI. Backup Power Requirements – Facilities-Based Providers of Non-Line Powered Residential Fixed Voice (Section 12.5 of the FCC's Rules)

Providers of the covered service (any facilities-based, fixed voice service offered as residential service that is not line powered) must offer at the point of sale the option to purchase backup power.

- Effective February 16, 2016: Providers with 100,000+ subscribers must offer at least one option for a minimum of eight hours of standby backup power.



- Effective August 11, 2016: Providers with fewer than 100,000 subscribers must offer at least one option for a minimum of eight hours of standby backup power.
- Effective February 1, 2017: All providers of the covered service must make certain backup power disclosures to new subscribers and to all annually
- Effective February 13, 2019: All providers of the covered service must offer at least one option for twenty-four hours of standby backup power.

These Section 12.5 rules have a sunset date of September 1, 2025, when they will no longer be effective.

Please contact John Kuykendall at jkuykendall@jsitel.com or Valerie Wimer at vwimer@jsitel.com or 301-459-7590 with any questions regarding interconnected VoIP compliance or for assistance with preparation of notices or filings.

DDM-5



State Universal Service Funds 2014

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Executive Summary

Universal Service is a key component of both Federal and State communications policy. Its goal is to ensure that all citizens have access to robust, reliable communications services, including broadband connectivity, at affordable rates, with "reasonably comparable service" across the country. Federal Universal Service funds (FUSF) provide a baseline for ensuring that comparable service is available to both urban and rural consumers. State funds both add to support provided by the Federal USF and are used to provide targeted support to address specific issues faced by each state's consumers.

NRRI's 2014 State USF review examines the way in which the states have addressed the question of universal service through state funds that supplement the four areas defined by the FCC--high cost support, low income support, support for schools and libraries (E-rate), and rural healthcare support. This paper examines changes to the state USF funds between 2012 and 2014 due to legislation, the FCC's USF Transformation Order, new rate benchmarks, and the move to include broadband in the Connect America Fund (CAF). The paper addresses the ways in which carriers and end users contribute to the funds, as well as the ways in which State funds are disbursed. This discussion provides data that may help State regulators and others respond to the FCC's current examination of the Federal USF contribution methodology, as well as manage their own State funds. The facts provided by the study will help the States make decisions on their funds, the FCC to understand the impacts of the ICC/USF Transformation Order on the states, and provide input on the way in which fund contributions may be structured in the future.

Forty-nine states and the District of Columbia responded to the NRRI 2014 survey.¹ Only one state, Hawaii, did not respond.

The states have multiple funds to support multiple universal service obligations. For simplicity, NRRI uses the term State USF in this study to refer to all of these funds, including access restructuring funds (Intrastate Access Support or IAS), Lifeline funds, Telecommunications Relay Service (TRS), accessible telecommunications equipment (TEP) funds to provide specialized customer premises equipment to the hearing and visually impaired, and other funds established by state law.

In all, 45 states provide some form of State universal service support in addition to the Federal funds. Six states, Alabama, Florida, Massachusetts, New Jersey, Tennessee, and Virginia, have no State funds. Although it has no fund, Florida requires all carriers to provide

¹ For simplicity, we refer to the District of Columbia as a state throughout this report.

Lifeline service. Massachusetts has no State fund but provides broadband support through a State grant program.

State USF support includes high cost support (22 states), funds for broadband access for schools and libraries (5 states), funding for Lifeline (17 states), and dedicated broadband funding (5 states). The majority of states direct USF contributions to specific funds. Two states, Texas and Washington, use a different methodology. Texas collects its USF as a single lump sum, which is then disbursed by the Commission to each state fund based on need. Washington funds universal service through the State's General Fund and then directs it to specific funds.

The largest proportion of SUSF funding (both in the number of states with a fund and the dollar value of that fund) is directed to supporting carriers that provide service in high cost or remote areas. Nearly half of the states with funds (22) provide high cost support. State high cost funds provide financial support for providers offering service in high cost and remote areas. Changes to the high cost funds over the study period, including the reduction or elimination of funding in areas served by competitive suppliers, have reduced the size of the fund in some cases or redirected monies to other uses in other cases.

Three states have Intrastate Access Restructuring Support (IAS) funds specifically designed to mitigate the effects of access charge reductions on carriers. For example, Michigan's fund is designed specifically to mitigate the effects of bringing intrastate access charges into alignment with interstate access charges on rural carriers. Where the states support IAS reform but do not designate a separate fund, we include their value in the high cost fund.

The State Universal Service funds grew just under 10% over the study period, from \$1,354,782,370 in 2012 to \$1,484,569, 879 in 2014. The growth in the funds was largely driven by significant increases in broadband and E-Rate funding in California and high cost growth in Illinois. The growth of State USF funds was tempered by reductions in Lifeline support and IAS funding, both driven by changes in federal regulation. State Lifeline funding decreased over the study period, as a result of both reductions in State support levels and more stringent eligibility requirements, including the elimination of duplicate registrations. One state, Wyoming, eliminated its State Lifeline program at the end of the study period. Additional reductions in Lifeline funding will occur over the next few years, as more states limit the amount of support they provide above the Federal benefit.

Contributors to the State USF vary by state and often by fund. All 50 of the states that responded to the NRRI survey assess wireline carriers, including CLECs. More than half of the states (32) assess intrastate long distance carriers (IXCs). Over half of the respondents (28) assess wireless providers. Seventeen states assess intrastate voice service provided by cable companies, while 13 states also assess interconnected VoIP providers.² Eight states assess end

² For the purposes of this paper, we categorize voice service provided by cable companies separately from other interconnected voice services, such as those provided by Vonage or Skype. AT&T's U-Verse service and Verizon's FiOS service are also included in the interconnected VoIP category.

users. Twelve states assess paging companies. In some states, cable and interconnected VoIP providers contribute voluntarily. Voluntary contributors include one VoIP provider in New York and one cable company in Utah, as well as some VoIP providers in Oregon. Unlike the Federal fund, which assesses providers a flat rate adjusted on a quarterly basis, collection by States differs depending on the fund to be supported. This allows the states to hone their funding requirements more specifically and to test out different contribution and funding methodologies.

The State Fund Overview table summarizes the findings of the 2014 NRRI Universal Service Survey.

State Universal Service programs are a significant tool for meeting the important policy goal of ensuring access to telecommunications for all citizens, regardless of where they live or their financial status. Continuing study and review of information on how various states meet this goal will remain an important public utility commission activity, now and in the future.

State Fund Overview

State	Who is Assessed?								On What Basis?			
	Landline	Wireless	VoIP	Cable	IXCs	Paging	End Users	Other	Intrastate Revenues		Per Line	Other
									Gross	Net		
AL												
AK	X	X			X	X	X		X			
AZ	X	X	X	X	X	X			X		X	
AR	X	X	X	X	X				X			
CA	X	X	X				X				X	
CO	X	X			X	X			X			
CT	X	X			X				X			
DC	X		X	X					X			
DE	X				X				X			
FL												
GA	X		X	X			X	X	X			
HI												
ID	X				X						X	X (4)
IL	X				X					X		
IN	X	X			X					X		
IA	X	X			X				X			X (5)
KS	X	X	X	X	X	X				X		
KY	X	X	X	X							X	
LA	X	X	X	X	X				X			
ME	X	X	X	X	X					X		
MD	X	X	X	X	X						X	
MA												

State	Who is Assessed?								On What Basis?			
	Landline	Wireless	VoIP	Cable	IXCs	Paging	End Users	Other	Intrastate Revenues		Per Line	Other
MI	X	X			X			X		X		
MN	X	X	X	X	X	X					X	
MS												
MO	X			X	X							X
MT							X				X	
NE	X	X	X	X	X	X				X		
NV	X	X	X	X						X		
NH	X			X							X	
NJ												
NM	X	X		X		X				X		
NY	X		X (1)	X						X		
NC							X				X	
ND	X	X									X	
OH	X	X	X	X	X	X						X
OK	X	X	X	X	X	X			X			
OR	X	X	X (2)		X		X		X		X	
PA	X				X			X		X		
RI							X				X	
SC	X				X		X					X (6)
SD	X	X			X	X	X				X	
TN												
TX	X	X			X	X			X			
UT	X	X		X (3)	X				X			
VT							X				X	
VA												
WA								X				X (7)
WV								X				
WI	X	X	X	X	X				X			
WY	X	X		X	X	X			X			

Notes:

(1) NY: One VoIP provider contributes voluntarily

(2) OR: VoIP providers contribute voluntarily

(3) UT: One cable company contributes voluntarily

(4) ID: Contribution differs by program

(5) IA: Wireless contribution by assigned number

(6) SC: Wireline: Retail rev., Relay per line, IAS allocated from prior year

(7) WA: Allocation from State general fund